SmartNode & SmartLink Voice over IP

Product Line Summary

Product Line	Series	Description	Pg
Residential	M-ATA	Micro-Analog Telephone Adapter	16
	M-AFA	Micro-Analog Fax Adapter	16
	S-ATA	Residential Smart Analog Telephone Adapter	17
	S-DTA	Residential Smart Digital BRI Telephone Adapter	19
	S-WTA	Residential Smart Wireless Analog VoIP IAD	18
Branch Office SOHO	SN411X	Multi-Port Analog VoIP Gateway	21
	SL402X	Analog VoIP SoHo Router	20
	SN455X	ISDN BRI VoIP SoHo Router	21
Enterprise – Large Enterprises – Medium Enterprises	SN452X	Multi-Port Analog VoIP IAD	22
	SN483X	Multi-Port Analog IAD with Integrated WAN Access	26
	SN463X	Multi-Port ISDN VoIP IAD	25
	SN465X	Multi-Port ISDN VoIP IAD with Integrated WAN Access	27
	SN4960	Multi-Port T1/E1 VoIP IAD	28
Carrier	SN4900	IpChannel Bank	29
	SN2400	4-Slot Modular VoIP Routers	30




								
Model	M-ATA, M-AFA	S-ATA	S-DTA	S-WTA	SL402X	SN411X	SN455X	SN452X
Description	Micro-Analog Telephone Adapter & Micro-Analog Fax Adapter	Smart Analog Telephone Adapter	Smart Digital (BRI) Telephone Adapter	Smart Wireless Analog VoIP IAD	Analog VoIP SOHO Router	Multi-Port Analog Gateway	ISDN BRI VoIP SOHO Router	Multi-Port Analog VoIP Router
VoIP Call Capacity	2	4	2	2	4	up to 8	2 / 4	up to 8
Ethernet Ports	1	2	1	5	2	2	5	2
Voice Interfaces	1	1 or 2	1	2	1	up to 8	2	up to 8
WAN Data Interfaces	-	-	-	-	-	-	-	-
Expansion Modules/Slots	-	-	-	-	-	-	-	-
Call Control	SIP	SIP	SIP & H.323	SIP	SIP or MGCP	SIP, H.323, and MGCP		
CODECs/Fax	G.711, G.729, G.726, G.723.1 T.38 & G.711 Fax bypass	G.711, G.729ab, G.726, G.723.1,T.38			G.711, G.729, G.726, G.723.1, T.38 & G.711 Fax bypass	G.711, G.723.1, G.729, G.729, G.729a, G.729b,G.729ab, G,727, G.726, T.38 with G3 Fax Relay; G.711 Fax Bypass		
Quality of Service	VLAN tagging and queuing, TOS and Diffserv labeling	TOS and DiffServ labeling	VLAN tagging and queuing, TOS and Diffserv labeling	TOS and DiffServ labeling	VLAN tagging and queuing, TOS and Diffserv labeling	TOS and DiffServ labeling	Same as SN483X	
IP Connectivity Features/VPN	DHCP client, DynDNS client, DNS SRV, VLAN, SNMP, SNTP, WWW,PPPoE, STUN, Syslog	IP Router, NAT/NAPT, DHCP server/client	IP Router, NAT/NAPT, DHCP client, DynDNS client, VLAN, SNTP, PPPoE	IP Router, NAT/NAPT, DHCP server/client	NAT, NAPT, DHCP client/ server, DynDNS client, DNS SRV, VLAN, SNMP, SNTP, PPPoE, STUN, SYSLOG	NAT/NAPT; DHCP server/client; DNS Relay; DynDNS client; SIP DNS SRV; VLAN .p/Q; SNMP, SNTP, WWW GUI, RIPv1/v2, PPPoE		

More Than Just Talk

Enterprise Telephony Products

Product Line	Series	Description	Pg
IP Phones 	SL4050/2	IP Phone with 2 lines	34
	SL4050/10	IP Phone with 10 lines	34
IP-PBX Appliances 	SIPxNano-15	Small Office IP-PBX for up to 15 extensions	32
	SIPxNano-30	Small Office IP-PBX for up to 30 extensions	32
	SL4250-75	Medium Office IP-PBX for up to 75 extensions	35
	SL4250-125	Medium Office IP-PBX for up to 125 extensions	35
	SL4250-250	Medium Office IP-PBX for up to 250 extensions	35

In This Section

						
Model	SN483X	SN463X	SN465X	SN4960	SN4900	SN2400
Description	Integrated WAN Access VoIP Router	Multi-Port ISDN VoIP IAD	Multi-Port ISDN VoIP IAD with Integrated WAN Access	Multiport T1/E1 VoIP IAD	IP Channel Bank	4-Slot Modular VoIP Gateway
VoIP Call Capacity	up to 8	4 / 8	4 / 8	up to 120	up to 32	up to 120
Ethernet Ports	2	2	2	2	2	2
Voice Interfaces	up to 8	3 / 5	3 / 5	4	up to 32	up to 16
WAN Data Interfaces	V.35, X.21, T1/E1, ADSL, G.SHDSL	-	V.35, X.21, T1/E1, ADSL, G.SHDSL	V.35, X.21, T1/E1, ADSL, G.SHDSL	V.35, X.21, T1/E1, ADSL, G.SHDSL	-
Expansion Modules/Slots	-	-	-	-	-	4-for BRI, PRI, T1/E1, or FXS
Call Control	SIP & H.323					
CODECs/Fax	G.711, G.723.1, G.729, G.729, G.729a, G.729b, G.729ab, G.727, G.726 T.38 with G3 Fax Relay; G.711 Fax Bypass					
Quality of Service	TOS and DiffServ labeling; Active QoS with traffic scheduling and classification. Weighted fair queuing and shaping of traffic classes with configurable tolerance; DownStreamQoS™ with dynamic restriction of inbound TCP traffic.					
IP Connectivity Features/VPN	NAT/NAPT; DHCP server/client; DNS Relay; DynDNS client; SIP DNS SRV; VLAN .p/Q; SNMP, SNTP, WWW GUI, RIPv1/v2, PPPoE Static Firewall ACLs; Filtering Ping DoS Detection; IPsec with DES/3DES/AES including Internet Key Exchange (IKE); VPN-Passthrough for PPTP/GRE					

SmartNode & SmartLink

Voice over IP

Gateway/Router Product Finder

	Number of Telephony Ports	Max Number of Simultaneous Calls	Telephony interfaces	Gateway or Router
Analog VoIP *From 1 to 32 ports for analog FXS or FXO IP connectivity* 	1	2	FXS	Gateway
	1 or 2	2	FXS	Router
	2 to 8	8	FXS and FXO	Gateway
	1 or 2	4	FXS	Router
	2 to 8	8	FXS and FXO	Router
	2 to 8	8	FXS and FXO	Router
	2	2	FXS	Wireless Router
	up to 32	up to 32	FXS	Router
ISDN BRI VoIP *The most comprehensive BRI VoIP solutions on the planet* 	1	2	BRI So	Gateway
	2	2 / 4	BRI So	Router
	3 / 5	4 / 8	BRI So	Router
	3 / 5	4 / 8	BRI So	Router
	4 to 16	32	BRI So	Router
T1/E1/PRI PSTN Gateway VoIP Trunking  *High-capacity trunk solutions*	1 or 4	15 to 120 software upgradeable	T1/E1/PRI	Router
Modular - Mixed Analog and Digital  *Do-anything VoIP boxes*	up to 16	120	T1/E1/PRI FXS, BRI	Gateway/Router

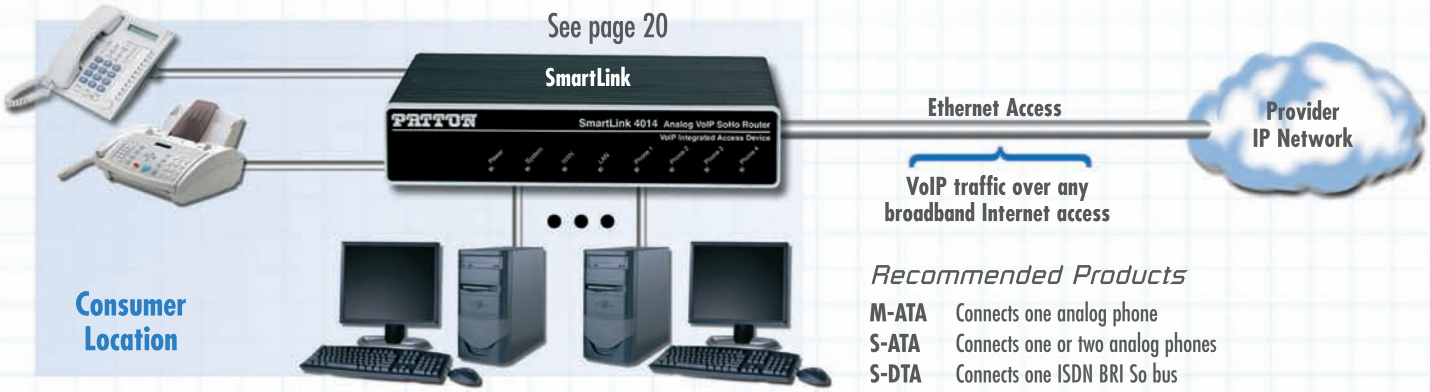
More Than Just Talk

Ethernet Ports	WAN Egress	Series	Series Description	Pg
1	10/100 Ethernet	M-ATA/M-AFA	Micro-Analog Telephone Adapter & Micro-Analog Fax Adapter	16
2	10/100 Ethernet	S-ATA	Residential Smart Analog Telephone Adapter	17
1	10/100 Ethernet	SN411X	Multi-Port Analog VoIP Gateway	21
2	10/100 Ethernet	SL402X	Analog VoIP SoHo Router	20
2	10/100 Ethernet	SN452X	Multi-Port Analog VoIP IAD	24
2	Ethernet, Sync. Serial, T1/E1, G.SHDSL or ADSL	SN483X	Multi-Port Analog IAD with Integrated WAN Access	26
5	10/100 Ethernet	S-WTA	Wireless Analog VoIP IAD	18
2	Ethernet, Sync. Serial, T1/E1, G.SHDSL or ADSL	SN4900	IpChannel Bank	29
1	10/100 Ethernet	S-DTA	Residential Smart Digital Telephone Adapter	19
5	10/100 Ethernet	SN455X	ISDN BRI VoIP SoHo Router/PSTN Gateway	23
2	10/100 Ethernet	SN463X	Multi-Port ISDN VoIP IAD	18
2	Ethernet, Sync. Serial, T1/E1, G.SHDSL or ADSL	SN465X	Multi-Port ISDN VoIP IAD with Integrated WAN Access	27
2	Ethernet	SN2400	4-Slot Modular VoIP Routers	30
2	10/100/1000 Ethernet	SN4960	Multi-Port T1/E1 VoIP IAD	28
2	Ethernet	SN2400	4-Slot Modular VoIP Routers/Gateways	30

SmartNode & SmartLink Voice over IP

Solutions Center Product Guide for Service Providers

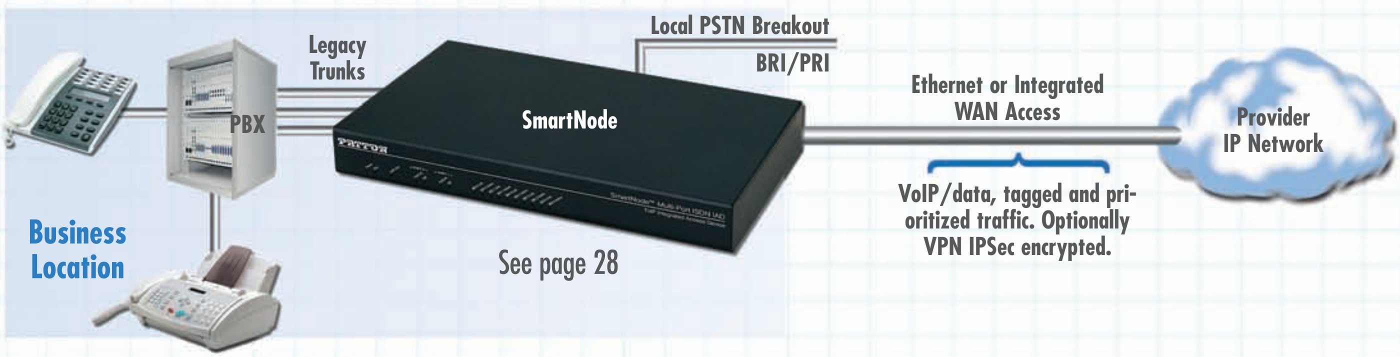
Residential VoIP Services



Recommended Products

M-ATA Connects one analog phone
S-ATA Connects one or two analog phones
S-DTA Connects one ISDN BRI So bus
S-WTA Connects two analog phones, provides wireless and wired access routing

Business Trunking

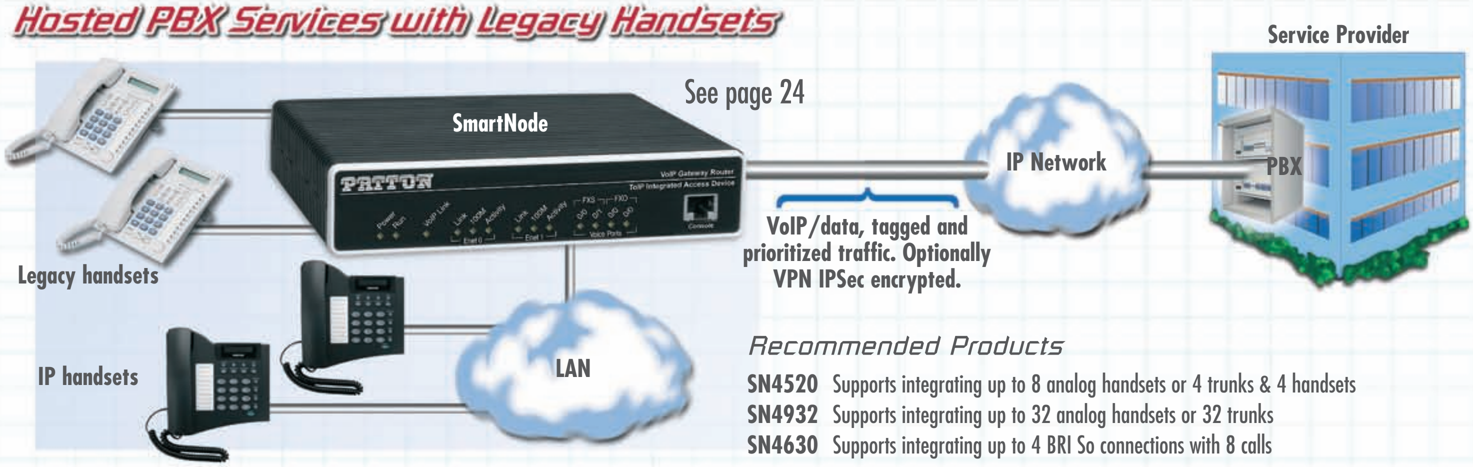


Recommended Products

SN4960 4 T1/E1/PRI ports to PBX, PSTN breakout capability, IP access routing with GigE, optional ADSL2+, G.SHDSL, X.21/V.35 or T1/E1 WAN access

SN4634 3 or 5 ISDN BRI So ports, 4 or 8 low-bandwidth voice or T.38 fax calls, PSTN breakout capability, Ethernet WAN access

Hosted PBX Services with Legacy Handsets



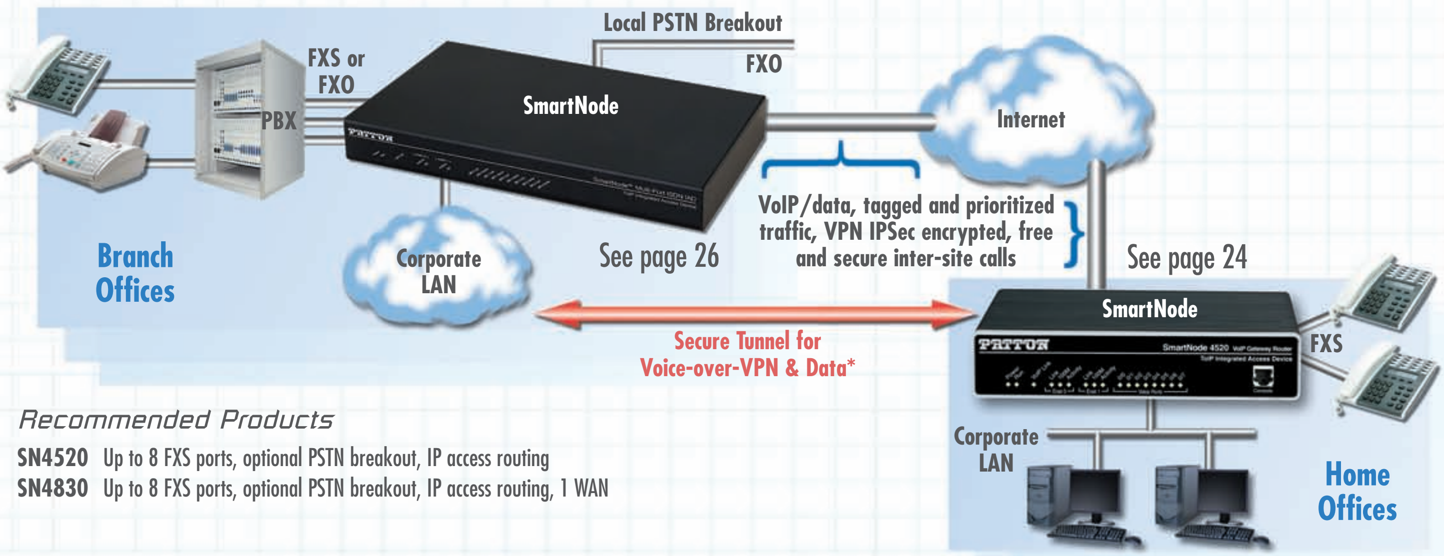
Recommended Products

SN4520 Supports integrating up to 8 analog handsets or 4 trunks & 4 handsets
SN4932 Supports integrating up to 32 analog handsets or 32 trunks
SN4630 Supports integrating up to 4 BRI So connections with 8 calls

More Than Just Talk

Solutions Center Product Guide for Enterprises

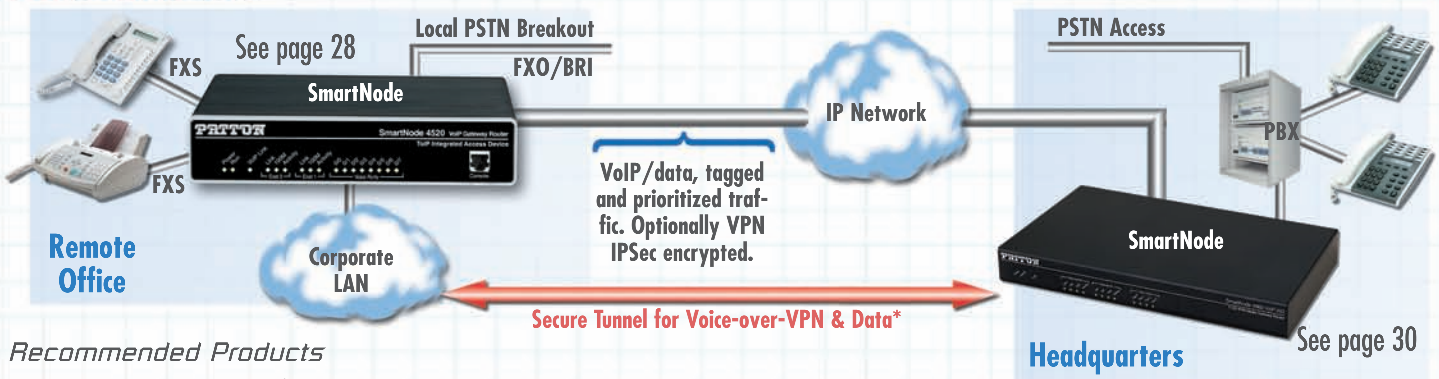
Branch Office Telephony



Recommended Products

SN4520 Up to 8 FXS ports, optional PSTN breakout, IP access routing
SN4830 Up to 8 FXS ports, optional PSTN breakout, IP access routing, 1 WAN

PBX Extension

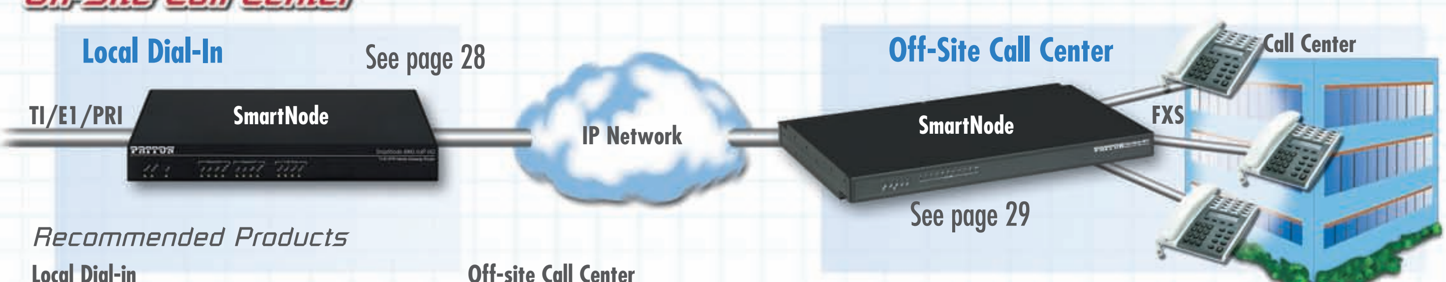


Recommended Products

SN4960/SN2400 Headquarters, for T1/E1/PRI PBX connection
SN4630 Headquarters, for BRI PBX connection
SN4520 Remote office, up to 8 analog line extensions

* Optional license sold separately

Off-Site Call Center



Recommended Products

Local Dial-in
SN4960 For T1/E1/PRI VoIP
SN4900 For analog dial-in — FXO

Off-site Call Center
SN4960 For T1/E1/PRI call center connection
SN4900 For analog call center connection — FXS

Micro-Analog Telephone Adapter & Fax Adapter

SmartLink™ M-ATA

Quickly and easily converts any phone or fax machine to VoIP for residential and telecommuter applications.

The SmartLink Micro Analog Telephone Adapter provides connectivity for analog phones and faxes to a home, home office or corporate LAN. Connecting to any analog phone, fax or PBX, the SmartLink product is a cost effective solution for small offices and telecommuters to access Internet-based telephone services and corporate intranet systems across established LAN and Internet connections like DSL and cable modems.

The M-ATA provides one Ethernet (RJ-45) port and one FXS (RJ-11) analog phone port for quick and easy interconnection to the local LAN. LEDs show at-a-glance the status of the system, LAN, WAN, and phone ports.

A full suite of IP features (DHCP) are available to maximize universal connectivity. VLAN tagging and prioritization enables voice traffic to be handled before data traffic, ensuring higher quality voice calls. Support for PPPoE tunneling simplifies extending corporate intranet services to telecommuters.

The user friendly web interface offers two levels of configuration: level one covers basic subscriber-specific parameters, level two offers advanced settings for the transport network. Configuration and firmware can be downloaded from a centralized TFTP or HTTP server.

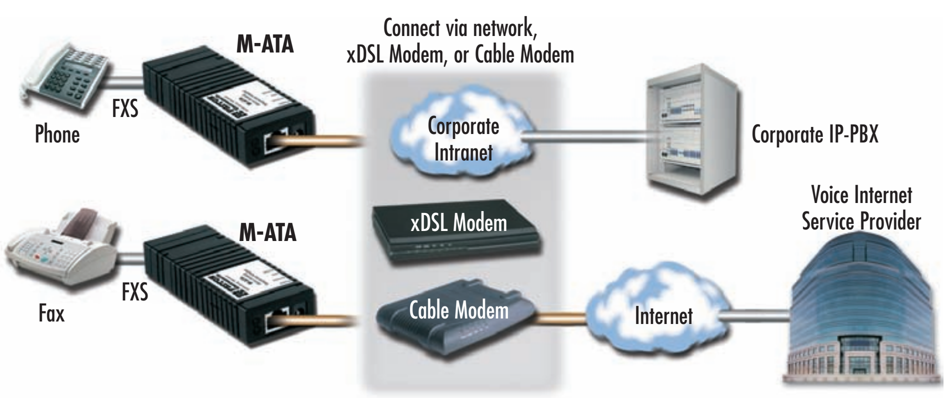
The M-ATA is SIP standard compliant. Analog phones attached to the SmartLink can use advanced calling features such as call forwarding, caller ID, 3-way calling, call holding, call retrieval, and call transfer.

VoIP-enable your existing fax machines

Integrated support for T.38 brings fax machines into the low-cost world of VoIP. The M-ATA supports G3 analog fax machines and converts fax signaling into T.38 or G.711 pass through fax for delivery over an Intranet or the Internet. Just configure the M-ATA, plug in the fax machine's phone line, add a LAN connection, and the M-ATA is ready to go!

Typical applications

Patton's Micro-Analog Telephone Adapter provides seamless access to Internet telephony and data services. The M-ATA connects to any broadband access provider via a cable or xDSL modem.



FEATURES & BENEFITS

- ✔ Ultra-miniature—Smallest full-function analog telephone adapter available today!
- ✔ Supports over 20 voice calling features—Call waiting, call conference, caller ID, hotline, distinctive ring and more!
- ✔ DHCP, PPPoE—Provides maximum connectivity across firewalls and transport networks.
- ✔ SIP Signaling—Deploy into any multimedia, interactive, or softswitch network with the leading call and session signaling protocols.
- ✔ Toll Quality CODECs & T.38 fax—Uses G.723 or G.729 for low-bandwidth applications or standard G.711 or G.726 CODECs for toll-quality voice.
- ✔ Centralized management—HTTP/SNMP manageable from any location.

ORDERING INFORMATION

M-ATA-1/E: Micro Analog Telephone Adapter; 1 x FXS RJ11; 1 x 10/100Base-TX

M-AFA-1/E: Micro Analog FAX Adapter; 1 x FXS RJ11; 1 x 10/100Base-TX, T.38 and G.711 only

SPECIFICATIONS

Voice Connectivity: 2-wire Loopstart, RJ-11/12 • Short haul loop 1.1 km @3REN • Caller-ID Type-1/2 FSK and ITU V.23/Bell 202 generation

Connectivity: 1 10/100Base-TX Full Duplex/Autosensing Ethernet RJ-45

Voice Processing (signalling dependent): SIP • Voice CODECS (G.711 A-Law/μ-Law (64 kbps); G.726 (ADPCM 40, 32, 24, 16 kbps); G.723.1 (5.3 or 6.3 kbps); and G.729ab (8 kbps)) • G.168 echo cancellation • 2 parallel voice connections • DTMF detection and generation • Carrier tone detection and generation • Silence suppression and comfort noise • Configurable dejitter buffer • DTMF in-band & out-of-band • Configurable transmit packet length • RTP/RTCP (RFC 1889) • STUN

Fax and Modem Support: G.711 transparent fax • T.38 fax relay (9.6 k, 14.4 k)

Voice Services/Features: Anonymous CallerID block • Call blocking • Call forward - on busy • Call forward - selective • Call forward - unconditional • Call hold/retrieve • Call return • Call transfer - blind • Call transfer - with consultation • Call waiting/retrieval • Caller-ID • Conference drop • Conferencing (3-way calling) • Distinctive ring • Do not disturb • Hotline calling • Incoming CallerID on/off • IP URL dialing • Message waiting indication • Self-caller ID block • Speed dial • Voicemail message retrieval • Warmline calling

IP Services: DHCP client • PPPoE • Programmable static routes • ICMP redirect (RFC 792); Packet fragmentation • VLAN support 802.1p/q

Management: Browser configuration interface • Multilevel security access • TFTP & HTTP configuration & firmware loading • SNMP v2 agent (MIB II and private MIB) • Syslog support

Operating Environment:
Op. temp.: 0–40°C (32–104°F)
Op. humidity: 5–80% (non-condensing)

System: Power: 100-240 VAC (50/60 Hz)

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 50950 • CE compliance • FCC Part 15 Class B

Physical:
Dimensions: 3.6L x 2.1W x 0.78H in. (9.0L x 5.3W x 1.9H cm)
Weight: 0.2 lbs (0.09 kg)

Smart-ATA Residential Analog Telephone Adapter

Model S-ATA

Leverage low-cost packet-voice and IP services for complete residential voice and data connectivity. The Smart-ATA supports full IP routing for transparent voice, fax, and data connections over any IP network.

The Smart Analog Telephone Adapter (S-ATA) provides VoIP connectivity for analog phones and faxes to the world of Internet voice. Connecting to any analog phone or fax, the Smart-ATA product is an effective and flexible solution for consumers and telecommuters to access Internet-based telephone services and corporate intranet systems across established LAN and Internet connections like xDSL and cable modems.

The S-ATA-A1 provides two RJ-45 Ethernet ports and one FXS (RJ-11) analog phone port. The S-ATA-A2 provides two RJ-45 Ethernet ports and two FXS (RJ-11) analog phone ports. Front panel LEDs quickly show at-a-glance the status of the system, LAN, WAN, and phone ports.

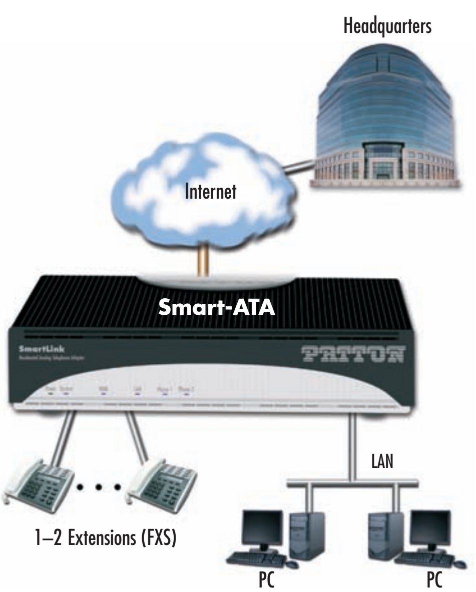
A full suite of IP features (DHCP, NAT/PAT, and NTP) are available to LAN devices attached downstream.

The web interface offers two levels of configuration access for the network operator and end user. Consumer friendly web interface and product labeling (Phone, LAN, WAN etc.) help ensure a trouble-free installation for the end user. Configuration and firmware can be downloaded from a TFTP server.

The Smart-ATA is SIP standard compliant, so it can be used with most SIP-based telephony services. Analog phones attached to the Smart-ATA can use such advanced voice calling features as call forwarding, caller ID, 3-way calling, call holding, call retrieval and call transfer.

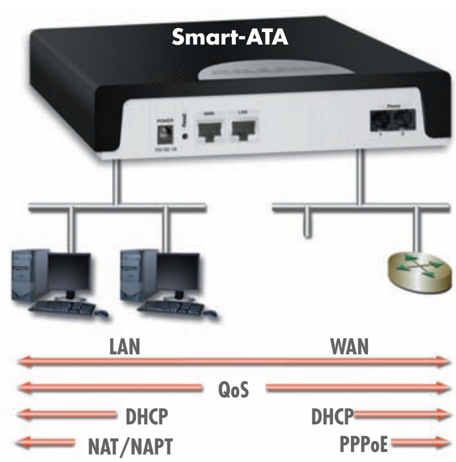
Local PSTN + Packet Voice

Patton's SmartLink Gateway routers with FXS support seamless access to Internet telephony and data.



LAN/WAN QoS & Router

As a premise router, the SmartLink offers voice and complete Internet access. With dual 10/100 Ethernet ports, the SmartLink connects your hosts to the LAN with VLAN tagging, DHCP server/client, and Firewall/ACL services. Use PPPoE and IPSEC VPN with DES, 3DES, and AES encryption and bring your Voice and Data to the WAN through a single and secure network connection.



FEATURES & BENEFITS

- ✔ Up to 2 FXS ports connect to your standard telephone or PBX.
- ✔ Quality of Service ensures voice traffic gets priority without shutting down your Ethernet LAN.
- ✔ NAT, DHCP, PPPoE—Connect to any broadband or access provider, serve the whole network, and secure your data. User configurable IP services ensure every host is connected to the LAN.
- ✔ SIP signaling—Deploy into any multimedia, interactive, or softswitch network with the leading call and session signaling protocols.
- ✔ Toll Quality CODECs & T.38 fax—Use standard G.711 or G.726 CODECs for toll-quality voice, or G.723 or G.729 for low-bandwidth applications.

ORDERING INFORMATION

S-ATA-A1/EUI/S: 1 Phone Port (FXS) Analog Telephone Adapter (ATA), SIP, UI power

S-ATA-A2/EUI/S: 2 Phone Port (FXS) Analog Telephone Adapter (ATA), SIP, UI power

I'm John, one of Patton's VoIP Products Group Managers. If you have technical questions or comments concerning VoIP Products, please call me at ***+1 (301) 975-1000, x182****. You can also send e-mail to* ***jwu@patton.com****.*

SPECIFICATIONS

Voice Connectivity: 2-wire Loopstart, RJ-11/12 • short haul loop 1.1 km at 3REN • Caller-ID type-1/2 FSK & ITU V.23/Bell 202 generation

Connectivity: 2 10/100 full duplex/autosensing Ethernet RJ-45

Voice Processing (signaling dependent): SIP • MGCP (Packet Cable NCS 1.0 & IETF MGCP 1.0) • Voice codes (G.711 A-Law/μ-Law (64kbps); G.726 (ADPCM 40, 32, 24, 16 kbps); G.723.1 (5.3 or 6.3 kbps); G.729ab (8kbps)) • G.168 echo cancellation • 4 parallel voice connections • DTMF detection & generation • Carrier tone detection & generation • Silence suppression & comfort noise • Configurable dejitter buffer • DTFM in-band & out-of-band • Configurable transmit packet length • RTP/RTCP (RFC 1889)

Fax and Modem Support: G.711 transparent fax • T.38 fax relay (9.6 k, 14.4 k)

Voice Services/Features: Call forwarding • Call transfer • Call hold • Call waiting • 3-way calling

IP Services: IPv4 router; RIPv1, v2 (RFC 1058 and 2453) • IP filtering • NAPT • NTP • DHCP client & server • programmable static routes • ICMP redirect (RFC 792); Packet fragmentation

Management: Browser configuration interface • TFTP configuration & firmware loading • SNMP v2 agent (MIB II and private MIB)

Environment: Temp: 32–104°F (0–40°C) • Humidity: 5–80% (non-condensing)

System: Power: 100–240 VAC (50/60 Hz) • Dimensions: 7.1W x 4.3D x 1.1H in. (18.0W x 11.0D x 2.7H cm) • Weight: 14.4 oz (236 g)

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 50950 • CE compliance • FCC Part 15 Class B

Smart Residential VoIP Analog WiFi Telephone Adapter

Model S-WTA

Take advantage of low-cost VoIP services without sacrificing existing Internet access and connectivity. The S-WTA offers dual channel VoIP, broadband routing and integrated WiFi for a complete residential TriplePlay Ready voice and data solution.

The Residential Smart WiFi Telephone Adapter (S-WTA) delivers connectivity with cost savings for residential and SoHo users. Combining two ports of standards-based VoIP with an integrated broadband WiFi router.

The S-WTA offers dual two-wire FXS phone ports for standard telephones or fax machines. With softswitch-certified SIP signaling, the S-WTA provides seamless toll-quality access to lowest cost VoIP services. The integrated broadband router allows you to connect to any Ethernet-based WAN such as ADSL, cable or fiber services. The integrated 4-port 10/100 LAN switch conveniently connects all of your networking devices. The integrated 802.11 WiFi offers b, g and super-g wireless connections automatically for highest throughput and maximum compatibility. Wireless WPA security with NAT, firewall/statefull packet inspection and denial-of-service support helps keep your network secure and your phone bills low.

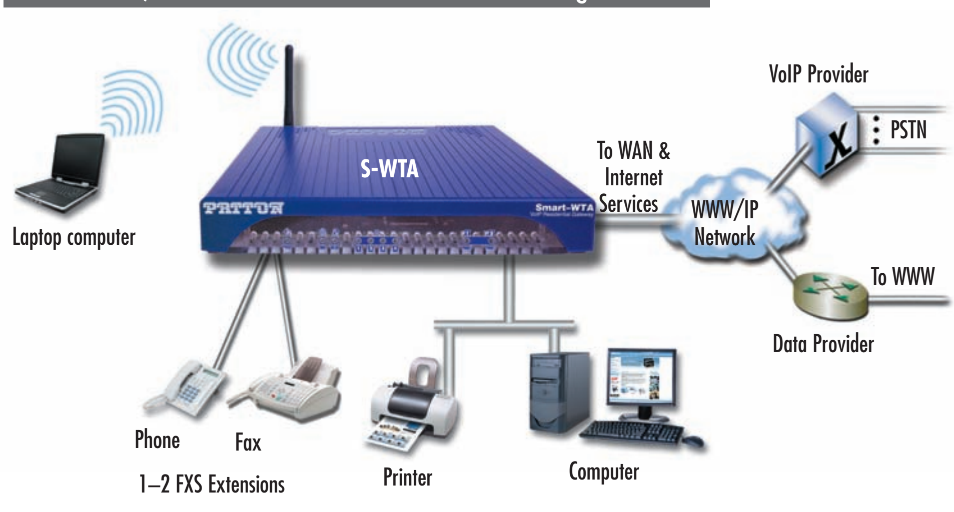
FEATURES & BENEFITS

- ✔ 2-call voice & fax FXS ports, 4-port LAN Ethernet switch, 1-port WAN Ethernet and integrated 802.11g WiFi
- ✔ SIP signaling—Deploy into any multimedia, interactive, or voice network with softswitch certified and tested call & session signaling.
- ✔ Integrated firewall/statefull inspection firewall, denial-of-service protection and NAT/DMZ. Wireless security offers WEP, WPA, WPA2, and 802.1x wireless security support.
- ✔ Complete TriplePlay multimedia and IPTV support with IGMP snooping and proxy. Integrated QoS for ensured voice, data, and multimedia quality.
- ✔ Toll quality and low-bandwidth CODECs. T.38 and fax-bypass features for solid multiuse interoperability.
- ✔ Single tightly integrated solution perfect for provider-based competitive calling and bundled data services.

ORDERING INFORMATION

S-WTA/G: 2-Port FXS VoIP Telephony Adapter with Broadband Router and Integrated 802.11b/g WiFi; 100–240 VAC external power supply (PS)

Remote Office/Branch Office Voice Extension and Access diagram



*I'm Denise, one of Patton's Sales Associates for Western Europe. Call me at **+1 301.975.1000** when you want to purchase Patton products or if you have questions about our products. You can also send e-mail to **sales@patton.com**.*

SPECIFICATIONS

Voice Connectivity: Two FXS 2-wire loopstart, RJ-11 • 5 REN • EuroPOTS (ETSI EG201 188) • programmable AC impedance, feeding, & ring voltage; on-hook voltage 50Vrms nominal• Caller-ID Type-1/2 FSK and ITU V.23/Bell 202 generation. Voice codes (G.711 A-Law/μ-Law (64kbps). Voice Processing: G.726 ADPCM; G.723.1 (5.3 or 6.3 kbps); G.729abe • G.168 echo cancellation • SIPv2 • DTMF detection & generation • carrier tone detection & generation • silence suppression & comfort noise • configurable dejitter buffer • configurable tones (dial, ringing, busy) • configurable transmit packet length • RTP/RTCP (RFC 1889)

Fax and Modem Support: G.711 transparent fax • Fax over IP (FoIP) • T.38 fax relay (9.6 k, 14.4 k)

IP Services: complete IPv4 router; DHCP server, PPPoE, NAT, DMZ with Configurable SPI Firewall and DoS protection• DiffServe/ToS set for queue per header bits • Pocket Policing discards excess traffic • 802.1p/Q VLAN support with 4096 IDs • AES/DES/3DES encryption options

Connectivity: WAN - Single 10/100 Ethernet RJ-45; LAN - Four RJ-45 switch ports; Integrated WiFi - dual diversity antenna. LAN switched with full country channel selection (US, Canada, Japan, Euro) at 2.4Ghz; autorate speed from 1-54 Mbps with Super-G burst support to 108 Mbps. Up to 23 dBm transmit power. data rate 54 Mbps.

Voice Routing—Session Router: Local switching; Interface hunt-groups • Routing Criteria

Management: HTTP/CLI with local console and remote Telnet/SSH access • TFTP configuration & firmware loading • SNMP v1 agent (MIB II and private MIB) • Built-in diagnostic tools (trace, debug)

Environment: Temp: 32–104°F (0–40°C) • Humidity: 5–80% (non condensing)

Power: 100–240 VAC (50/60 Hz) • Power dissipation: 4W

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 50950 • CE compliance • FCC Part 15 Class A

Smart-DTA Residential Digital Telephone Adapter

Model S-DTA

The S-DTA brings all advantages of VoIP to ISDN users. It connects multiple terminals to its BRI So bus, and converts 2 concurrent voice or fax calls to SIP or H.323 —at an incredibly low price.

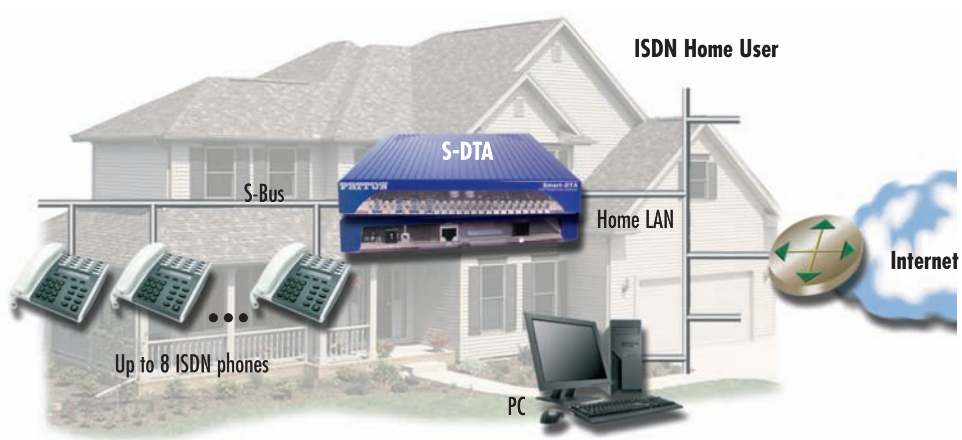
The Smart-DTA enables integration of ISDN network users into a local VoIP phone service, or extends an ISDN line of a PBX to a remote site over IP. It offers a simple end-user configuration interface and connects both to a PBX in point-to-point mode and an So bus in point-multipoint mode.

Unlike most other products on the market, Patton's intelligent call routing technology does not only offer simple ISDN to VoIP, but also advanced features like number plan adaptations, mappings between ISDN and SIP/H.323, manipulation of call properties through regular expressions, routing calls based on time-of-day or bearer capability criteria and much more.

Providing power to the ISDN line, the S-DTA eliminates the need for an external power supply to provide power to the terminals. Gateway functions use standard CODECs such as G.723, G.729, and T.38 fax as well as industry standard SIP, H.323 and MGCP/IUA signaling protocols to ensure seamless connection and compatibility for all voice services. Quality of service (QoS) features complete the offering with advanced voice prioritization and traffic management including VLAN and 802.1p/Q tagging.

Network Integration

Patton's Smart Digital Telephony Adapter provides seamless access to Internet telephony services for ISDN terminals. The S-DTA connects to any LAN or broadband access provider via modem.



FEATURES & BENEFITS

- ✔ Instant ISDN to VoIP Connectivity—Provides one internally powered So bus for up to 8 terminals. Simultaneously converts 2 voice and fax calls to VoIP.
- ✔ Transparent Telephony Features—Preserves ISDN features like caller ID and name (CLIP/CLIR), call transfer, hold, waiting, charging information (AOC) and much more.
- ✔ Outstanding Interoperability—Interoperable for voice and fax calls with a wide variety of ISDN terminals, PBXs, as well as SIP and H.323 soft switches and application servers.
- ✔ Full SIP and T.38 support—Complete range of industry standard signaling protocols supported: SIPv2, H.323v4, MGCP/IUA, ISDN, BRI, T.38, fax and modem bypass, DTMF relay
- ✔ Integrated Management and Provisioning—Web GUI for easy end-user setup, fully automated provisioning system and SNMP management for mass deployment

I'm Michael, Associate Product Manager for VoIP. If you have technical questions or comments concerning VoIP Products, please call me at ***+1 301.975.1000, x202****. You can also send e-mail to* ***mhuarca@patton.com****.*

SPECIFICATIONS

WAN Connectivity: 10/100Base-T Ethernet WAN • Auto-MDI-X • DHCP Client • PPPoE Client (multi-session) • SNTP • IP Multi-Netting

IP Quality of Service: IEEE 802.1p, TOS, DiffServ Labeling • IEEE 802.1Q, VLAN Tag insertion/deletion (4,096 VLANs)

Management: Web-based GUI • Fully documented CLI • Telnet and HTTP access • TFTP configuration up- and download • TFTP firmware upgrade • SNMPv1 agent, MIB II and enterprise MIB • Built-in diagnostic tools • Auto-Provisioning—Configuration and Firmware

Fax and Modem Support: T.38 fax over IP • Fax relay and bypass • Modem bypass

ISDN Specification: 1 port Euro-ISDN BRI/So RJ-45, NT • DSS-1, Q.921, Q.931 • Point-point & point-multipoint

Voice Signaling: SIPv2 • H.323v4 • MGCP/IUA • SIP call transfer, redirect • Overlap or en-bloc dialing • DTMF in-band & out-of-band • Configurable call progress tones

Call Routing & Services: Regular expression number matching • Regular expression number manipulation • Least Cost Routing • Number blocking • Short-Dialing • Digit collection • Distribution-Groups & Hunt-Groups • 2nd call offering

Voice Processing: G.723.1 (5.3/6.3 kbps) • G.729, G.729a, G.729ab (8 kbps) • G.726 ADPCM (16 ,24, 32, 40 kbps) • G.168 echo cancellation (25ms) • Transparent ISDN data • Silence suppression and comfort noise • Adaptive and configurable dejitter buffer • Configurable packet length

Power & Packaging: Dimensions: 4.2W x 1.5H x 5.0D in. (10.6W x 3.9H x 12.7D cm) • Weight: < 15.9 oz (450 g) • Power Consumption < 4W

Environment: Temp.: 32–104°F (0–40°C) • Humidity: up to 90%, non condensing

Compliance: FCC Part 15 Class B (US EMC) • CE per RTTE 99/5/EC (EMC and LVD) • Safety—EN60950 • TBR-3 (ISDN BRI/So)

ORDERING INFORMATION

S-DTA/EUI: ISDN VoIP adapter, 1x BRI/So, 2 voice/fax calls, 1x 10/100 Ethernet, external UI power (100–240 VAC)

Analog VoIP SoHo Router

SmartLink™ 4020 Series

The SmartLink 4020 VoIP SoHo Router connects your LAN, standard analog phones, and fax to any IP network. This full-featured IP router includes VPN/Security and Quality of Service for converged low-cost sales-office/telecommuter voice data communications.

The SmartLink VoIP SoHo Router provides transparent connectivity for analog phones and faxes to the world of Internet voice. Connecting to any analog phone, fax or PBX, the SmartLink product is an effective and flexible solution for small offices and telecommuters to access Internet-based telephone services and corporate intranet systems across established LAN and Internet connections like xDSL and cable modems.

The SmartLink Model 4021 provides two RJ-45 Ethernet ports and one FXS (RJ-11) analog phone port. The SmartLink Model 4022 provides two RJ-45 Ethernet ports and two FXS (RJ-11) analog phone ports. Front panel LEDs quickly show at-a-glance the status of the system, LAN, WAN, and phone ports.

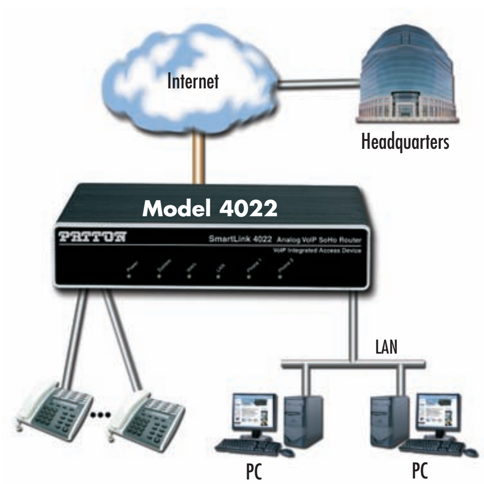
A full suite of IP features (DHCP, NAT/PAT, NTP and VPN) are available to LAN devices attached downstream. VLAN tagging and prioritization enables voice traffic to be handled before data traffic. Support for PPPoE and IPSEC tunneling simplifies extending corporate intranet services to remote teleworkers.

The web interface offers two levels of configuration access for the network operator and end user. The friendly web interface and product labeling (Phone, LAN, WAN etc.) to help ensure a trouble-free installation for the end user. Configuration and firmware can be downloaded from a TFTP server.

The SmartLink is SIP standard compliant, so it can be used with most SIP-based telephony services. Analog phones attached to the SmartLink can use such advanced voice calling features as call forwarding, caller ID, 3-way calling, call holding, call retrieval and call transfer.

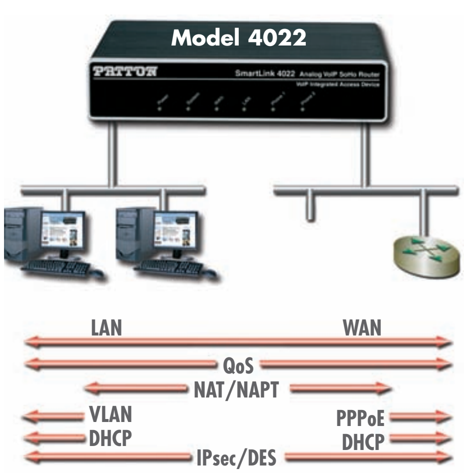
Local PSTN + Packet Voice

Using Patton's SmartLink VoIP SoHo routers with FXS supports seamless access to Internet telephony and data.



LAN/WAN QoS & Router

As a premise router, the SmartLink offers voice and complete Internet access. With dual 10/100 Ethernet ports, the SmartLink connects hosts to your LAN with VLAN tagging, DHCP server/client, and Firewall/ACL services. Use PPPoE and IPSEC VPN with DES, 3DES, and AES encryption and bring your voice and data to the WAN through a single and secure network connection.



FEATURES & BENEFITS

- ✔ Up to 2 FXS ports connect to a standard phone or PBX.
- ✔ Quality of Service ensures voice traffic gets priority without shutting down your Ethernet LAN.
- ✔ Firewall, NAT, DHCP, PPPoE—Connect to any broadband access provider, serve the whole network, and secure your data. User configurable IP services ensure every host is connected to the LAN.
- ✔ SIP Signaling—Deploy into any multimedia, interactive, or softswitch network with the leading call and session signaling protocols.
- ✔ Toll Quality CODECs & T.38 fax—Uses standard G.711 or G.726 CODECs for toll-quality voice, or G.723 or G.729 for low-bandwidth applications.

ORDERING INFORMATION

SN4021/EUI/S: 1 Port FXS VoIP Gateway Router, 100–240 VAC external power supply (PS), SIP

SN4021/EUI/M: 1 Port FXS VoIP Gateway Router, 100–240 VAC external PS, MGCP

SN4022/EUI/S: 2 Port FXS VoIP Gateway Router, 100–240 VAC external PS, SIP

SN4022/EUI/M: 2 Port FXS VoIP Gateway Router, 100–240 VAC external PS, MGCP

SPECIFICATIONS

Voice Connectivity: 2-wire loop-start, RJ-11/12 • short haul loop 1.1 km @3REN • Caller-ID Type-1/2 FSK & ITU V.23/Bell 202 generation

Voice Processing (signaling dependent): SIP • MGCP (Packet Cable NCS 1.0 & IETF MGCP 1.0) • Voice CODECs (G.711 A-Law/μ-Law (64 kbps); G.726 (ADPCM 40, 32, 24, 16 kbps); G.723.1 (5.3 or 6.3 kbps); G.729ab (8 kbps)) • G.168 echo cancellation • 4 parallel voice connections • DTMF detection & generation • carrier tone detection & generation • silence suppression & comfort noise • configurable dejitter buffer • DTFM in-band & out-of-band • configurable transmit packet length • RTP/RTCP (RFC 1889)

Voice Services/Features: Call forwarding • Call transfer • Call hold • Call waiting • 3-way calling

IP Services: IPv4 router; RIPv1, v2 (RFC 1058 & 2453) • IP filtering • NAPT • NTP • DHCP client & server • PPPoE • IPSEC VPN • programmable static routes • ICMP redirect (RFC 792); Packet fragmentation • DiffServe/ToS set or queue per header bits • VLAN support 802.1p/q • AES/DES/3DES encryption

Fax and Modem Support: G.711 transparent fax & T.38 fax relay (9.6 k, 14.4 k)

Connectivity: 2 10/100 Full Duplex/Autosensing Ethernet RJ-45

Management: Browser configuration interface • TFTP configuration & firmware loading • SNMP v2 agent (MIB II & private MIB)

Environment: Temp.: 32–104°F (0–40°C) • Humidity: 5–80% (non condensing)

System: Power: 100–240 VAC (50/60 Hz)

Compliance: EMC compliance: EN55022 & EN55024 • Safety compliance: EN 50950 • CE compliance • FCC Part 15 Class B

Multi-Port FXS/FXO VoIP Gateway Series

SmartNode™ 4110 Series

The SmartNode 4110 VoIP gateway integrates up to eight legacy PSTN, PBX or standard phone lines with next-generation IP based telephony systems. Part of the proven SmartNode family, this product is designed to provide superior technology at an optimized cost.

The SmartNode 4110 VoIP Media Gateway supports up to eight transparent phone calls while leveraging VoIP for lower-cost carrier and corporate access. Connecting to any analog phone, fax, or PBX, the SN4110 is an effective and flexible solution for toll-bypass, remote/branch office voice connectivity, and enhanced carrier services.

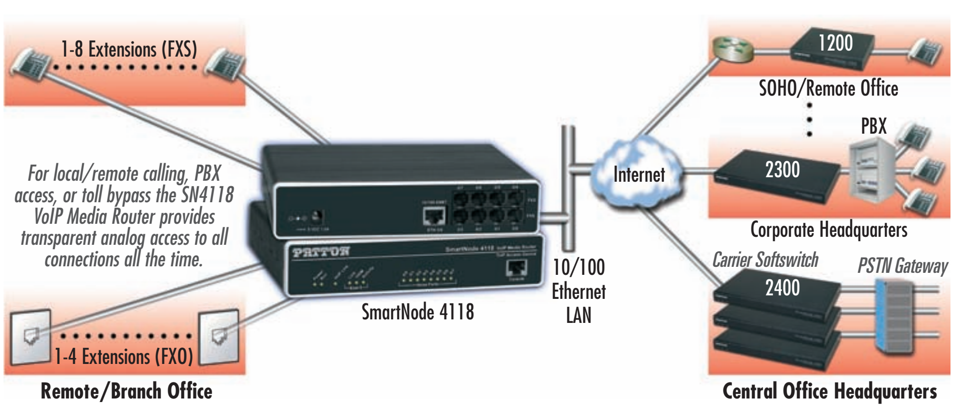
The SN4110 series is the perfect choice for phone-to-IP connectivity supporting up to 8 FXS ports or a combination of 4 FXS and 2 or 4 FXO ports. With its FXS analog ports, the SN4110 connects to any legacy telephone or PBX and provides dial-tone, ringing, and caller-ID. When equipped with FXO ports, the local PSTN can be accessed enabling local calling and enhanced toll-bypass applications while using a single connected telephone. Flexible call integration allows per-port telephone numbers, programmable call progress tones, and distinctive ringing. With Telephony-over-IP (ToIP) call switching, calls can automatically select the least-cost-route while providing flexible numbering plans and end-to-end feature transparency. PPPoE, DHCP, and VLAN offers universal IP connectivity and optional IPSEC VPN with AES/3DES guarantees secure voice over the public network.

Patton's SmartNode 4110 delivers the legacy phone interfaces, service transparency, and flexible PSTN integration required for true converged packet voice.

FEATURES & BENEFITS

- ✔ Up to 8 FXS and/or FXO ports—Compact, reliable stand-alone VoIP gateway with different port options. Supports simultaneous voice or fax calls on all ports.
- ✔ Advanced Local Call Switching—Virtual interfaces and routing tables provide industry leading flexibility in call handling programming. Local call switching, soft fallback to alternative routes. Simultaneously connects to multiple SIP services/IP PBXs.
- ✔ Complete SIP and T.38 support—Supports the complete range of industry standard VoIP: SIP, H.323, T.38 fax, fax and modem handling, DTMF relay. Codecs G.729, G.723, etc.
- ✔ Easy Management & Provisioning—Web-based management, SNMP, command line interface. Automated mass provisioning for efficient large-scale deployments.
- ✔ Outstanding Interoperability—Proven integration for voice and T.38 fax with Asterisk™, PingTel™ and other leading IP PBX systems and soft switch vendors.

Remote Office/Branch Office Voice Extension and Access diagram



ORDERING INFORMATION

- **SN4112/JS/EUI**: 2 Port FXS VoIP Media Gateway, 100–240 VAC external power supply (PS)
- **SN4114/JS/EUI**: 4 Port FXS VoIP Media Gateway, 100–240 VAC external PS
- **SN4116/JS/EUI**: 6 Port FXS VoIP Media Gateway, 100–240 VAC external PS
- **SN4118/JS/EUI**: 8 Port FXS VoIP Media Gateway, 100–240 VAC external PS
- **SN4114/2JS2JO/UI**: 2 FXS & 2 FXO Port VoIP Media Gateway, 100–240 VAC internal PS
- **SN4118/4JS4JO/EUI**: 4 FXS & 4 FXO Port VoIP Media Gateway, 100–240 VAC external PS

Options & Accessories

SNSW-VPN1: License Key for IPSec VPN support (DES, 3DES, AES)

SPECIFICATIONS

Capacity: Up to 8 simultaneous VoIP or T.38 fax calls (depending on the model)
Voice Signaling: H.323v4, SIPv2 (B2BUA capable, multi-instance, simultaneous support of multiple registrars and direct IP dialing) • SIP call transfer, redirect • DTMF in-band & out-of-band • All tones programmable (dial, ringing, busy)
Voice Processing: CODEC G.711 a-law/mu-law, G.723, G.729ab, • G.726, G.727. T.38 fax relay (9.6 k, 14.4 k) • G.711 transparent fax and bypass
Call Switching and Services: Virtual interfaces • Regular expression based call routing and number manipulation • Number blocking • Short-dialing • Digit collection, distribution and hunt groups • Transparent line extension • Fallback Routing: Soft fallback to alternative route(s)
FXS Connectivity: 2-wire Loopstart on RJ-11/12 •short haul loop 1.1km @3REN • EuroPOTS (ETSI EG201188) • programmable AC impedance, feeding, ring and on-hook voltage • Caller-ID FSK and ITU V.23/Bell 202 generation
FXO Connectivity: 2-wire Loopstart on RJ-11/12 • Programmable impedance, ring detection, tone detection, disconnect supervision • Caller ID detection
IP services: One 10/100 Ethernet port • DHCP Client • access control lists • Traffic policing • IEEE 802.1p, TOS, DiffServ labeling • IEEE 802.1Q, VLAN tag insertion/deletion (simultaneous support of multiple VLANs) • IPSEC, IKE, AES/DES/3DES Encryption (optional)
Management: Web/HTTP, CLI with local console and remote Telnet access • TFTP configuration & firmware loading • SNMP MIB II and product MIB • Secure Mass provisioning for both firmware and unit/subscriber configuration • Built-in diagnostic tools (trace, debug, call generator)
System: CPU Motorola MPC870 @ 66MHz • Memory 32MB SDRAM/8MB Flash • Power 100–240 VAC (50/60 Hz) • Power dissipation 4-12W, model dependent
Environment:
Temp.: 32–104°F (0–40°C)
Humidity: 5–80% (non condensing)
Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 60950 • CE compliance • FCC Part 15 Class A • TBR21 (FXS) • RoHS

Small & Branch Office ISDN VoIP Routers

SmartNode™ 4552 (Standard) & 4562 (Secure)

The SmartNode 4552 and 4562 are the perfectly integrated devices to take branch office connectivity to the next level. Combining VoIP, ISDN and IP access routing, they enable the cost savings of VoIP while preserving speech and fax quality and reliability.

The SmartNode 4552 and 4562 enable the integration of ISDN branch offices or remote users into corporate voice and data networks. They attach any standard ISDN telephone or PBX to a public or corporate VoIP service, generating lots of communication cost savings without having to sacrifice quality.

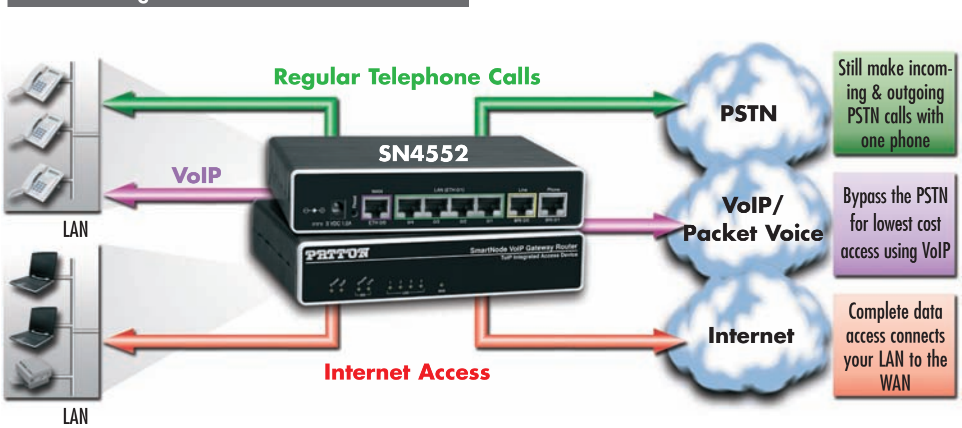
Industry-leading call switching features include hard and soft communication fallback to the ISDN breakout port in the event of failure, ensuring that no services are interrupted when migrating to VoIP.

The SmartNode 4562 adds hardware accelerated encryption power to the impressive feature list of the SmartNode 4552, making VoIP accessible to organizations that have been missing out on the cost-saving benefits of Internet telephony because of security concerns.

With the ClearConnect™ dial-backup option, adaptive network monitoring recognizes WAN uplink failures and initiates an ISDN dial-up connection to guarantee interrupt-free voice and data access at all times. If the VoIP link goes down or becomes congested, the SN4552 will switch over to the PSTN and guarantee your call each time.

Broadband network connectivity integrates with any fixed IP, DHCP or PPPoE service. An integrated 10/100 Ethernet LAN switch, with advanced routing features such as multiple VLANs, NAT, Firewall/ACL or DynDNS fulfills the requirements of demanding network users. Quality of Service (QoS) features complete the offering with advanced voice prioritization and traffic management. Patton's patent-pending DownStreamQoS™ ensures voice without interruptions even over best-effort Internet connections.

Network Integration



FEATURES & BENEFITS

- ✔ Full SIP and T.38 support—Complete range of industry standard signaling protocols supported: SIPv2, H.323v4, MGCP/IUA, DSS1, Euro-ISDN, VN4, T.38 fax, fax and modem bypass, DTMF relay.
- ✔ Toll-Quality VoIP—Advanced traffic management and shaping, combined with Patton's patent-pending DownStream QoS™ enforce uninterrupted toll-quality voice over best-effort networks.
- ✔ Transparent Telephony Features—Preserves ISDN features like caller ID and name (CLIP/CLIR), call transfer, hold, waiting, AOC and much more. Handles complex number manipulation for most seamless integration with existing infrastructure.
- ✔ Management & Provisioning—Built-in Web based management, SNMP, Command Line Interface and Auto-Provisioning for automated configuration distribution and software upgrades.
- ✔ Accelerated Voice over VPN*—Encrypts voice, signaling and data traffic over IP networks with IPsec, AES, 3DES and IKE. Complete access router with NAT, firewall, PPPoE, DHCP and DynDNS.
- ✔ ClearConnect™ dial-backup option for survivable voice and data connectivity.

ORDERING INFORMATION

SN4552/2BIS/EUI: ISDN BRI VoIP SoHo Router, 2-port, 10/100Base-T WAN, 4-port 10/100Base-T switch 100–240 VAC external power supply

SN4562/2BIS/EUI: SmartNode ISDN Voice over VPN Router, 2 port, 10/100Base-T WAN, 4-port 10/100Base-T LAN switch, 100-240 VAC external power supply. Includes VPN license key for IPSec VPN, IKE and Voice-Over-VPN. Hardware accelerated cryptography.

Options & Accessories

PM-BRI-EXT: External S-Bus Phantom-Power Supply 40 VDC for Phone only (required to provide power to telephones)

SNSW-QSIG1: License Key for QSIG support

SNSW-DB1: Dial-Backup Feature License

* Available on SN4562 only

SPECIFICATIONS

Capacity: 2 simultaneous VoIP or T.38 fax calls
ISDN Connectivity: 2 ports Euro-ISDN BRI/So RJ-45 • 1 NT port/1 TE port • DSS-1, Q.921, Q.931 • Point-point and point-multipoint • Lifeline Bypass Relay TE port to NT port • Optional QSIG support
VoIP Signaling: SIPv2 (B2BUA capable, multi-instance, supports registrar and direct IP dialing at the same time) • H.323v4 • MGCP/IUA • SIP call transfer, redirect • DTMF in-band & out-of-band • All tones programmable (dial, ringing, busy) • Overlap or en-bloc dialing• Transparent AOC, ECT, CLIP, CLIR, etc • speech, audio & data (Fax Gr 4, UDI 64, • RDI 64);
Voice Processing: CODEC G.711 a-law/mu-law, G.723, G.729ab, G.726, G.727. T.38 fax relay (9.6 k, 14.4 k) • G.711 transparent fax and bypass
LAN and IP Services: 4-port LAN Switch • Auto-MDI-X • IPv4, RIPv2, ICMP • Dynamic and static NAT and NAPT • ACL Firewall • DNS, DynDNS • DHCP Server • SNTP Client• IPSEC, IKE, AES/DES/3DES Encryption (hardware accelerated, on model 4562 only)
WAN Connectivity: 10/100BaseT Ethernet WAN • Auto-MDI-X • DHCP Client • PPPoE Client (multi-session) • IP Multi-Netting
Quality of Service: Voice priority • DownStreamQoS™ • Traffic management, shaping and policing • IEEE 802.1p, TOS, DiffServ labeling • IEEE 802.1Q, VLAN tag insertion/deletion (simultaneous support of multiple VLANs and PPPoE sessions)
Management: Web/HTTP, CLI with local console and remote Telnet access • TFTP configuration & firmware loading • SNMP MIB II and product MIB • Secure mass provisioning for both firmware and unit/subscriber configuration • Built-in diagnostic tools (trace, debug, call generator)
System: CPU Motorola MPC870 @ 66MHz • Memory 32MB SDRAM/8MB Flash • Power 100–240 VAC (50/60 Hz) • Power dissipation 4-12W, model dependent
Environment: Temp.: 0–40°C • Humidity: 5–80% (non condensing)
Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 60950 • CE compliance • FCC Part 15 Class A • TBR3 (ISDN) • RoHS

ISDN BRI PSTN Gateway

SmartNode™ 4554

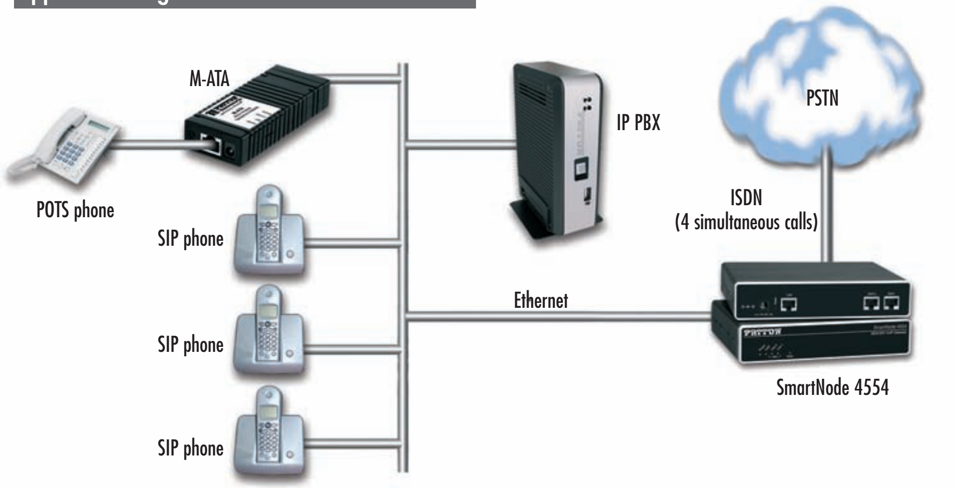
The SmartNode™ 4554 ISDN VoIP PSTN gateway enables any IPBX or IP phone system to seamlessly connect to the ISDN PSTN. The compact, reliable design allows easy and flexible integration of BRI lines into any VoIP system.

The SmartNode™ 4554 ISDN BRI PSTN Gateway converts up to four simultaneous phone or T.38 fax calls from SIP or H.323 to BRI ISDN. It is a compact reliable stand-alone VoIP gateway for IP-based voice systems that delivers ISDN performance and quality.

For system integrators looking for a VoIP solution with ISDN connectivity, the SmartNode 4554 provides unparalleled ISDN to IP preservation. The SmartNode provides advanced ISDN functionality such as Explicit Call Transfer (ECT) support, and Advice of Charge (AOC) over SIP. Patton CPEs are also interoperable with several leading IP PBX vendors such as Asterisk, 3CX, and SIP Foundry.

Compared to PC-based solutions that combine IP PBX and a PCI BRI cards, the SmartNode has several advantages, including: installation without additional drivers or software, operation without ventilation or hard disk, scaleable without limitation, and service integration without downtime. SmartNodes can be combined in a cluster for simple and flexible redundancy solutions not available with PCI BRI cards. System administrators don't have to learn ISDN as industry standard protocols like SIP and H.323 do the job of connecting the PSTN to your VoIP system.

Application diagram



I'm Jen, one of Patton's Sales Associates. Call me at ***+1 301.975.1000*** *when you want to purchase Patton products or if you have questions about our products. You can also send e-mail to* ***sales@patton.com****.*

FEATURES & BENEFITS

- ✔ Four Simultaneous SIP or T.38 calls—Connects an IP PBX or other VoIP systems to the ISDN PSTN with 2 BRI TE ports and 4 simultaneous voice or fax calls.
- ✔ Full VoIP Support—Supports the complete range of industry standard VoIP: SIP, H.323, T.38 Fax, Fax and modem bypass, DTMF relay, Codecs G.729, G.723, and G.711.
- ✔ Transparent Telephony Features—Preserves ISDN features like caller ID and name (CLIP/CLIR), call transfer, hold, waiting, ECT, AOC. Supports DDI and MSN lines in all major countries as well as complex numbering plan manipulations.
- ✔ Outstanding Interoperability—Interoperable for voice and T.38 fax with Asterisk™, 3CX™, PingTel™ and other leading IP PBX systems.
- ✔ Management & Provisioning—Built-in Web-based management, SNMP, Command Line Interface and Auto-Provisioning for automated configuration distribution and software upgrades.

ORDERING INFORMATION

SN4554/2BIS/EUI: ISDN BRI VoIP Gateway, 2 BRI ports, 1 Ethernet port, 4 simultaneous calls, 100–240 VAC external power supply

Options & Accessories

SNSW-VPN1: License Key for IPSec VPN support (DES, 3DES, AES)

SNSW-QSIG1: License Key for QSIG support

SPECIFICATIONS

WAN Connectivity: 10/100Base-T Ethernet WAN • Auto-MDI-X • DHCP Client • PPPoE Client (multi-session) • SNTP • IP Multi-Netting

IP Quality of Service: IEEE 802.1p, TOS, DiffServ Labeling • IEEE 802.1Q, VLAN Tag insertion/deletion 4,096

Management: Web-based GUI • Fully documented CLI • Telnet and HTTP access • TFTP configuration up- and download • TFTP firmware upgrade • SNMPv1 agent, MIB II and enterprise MIB • Built-in diagnostic tools • Auto-provisioning—configuration and firmware

Fax and Modem Support: T.38 fax over IP • Fax relay and bypass • Modem bypass

ISDN Specification: 2 port Euro-ISDN BRI/So RJ-45, TE • DSS-1, Q.921, Q.931 • Point-point & point-multipoint

Voice Signaling: SIPv2 • H.323v4 • MGCP/IUA • SIP call transfer, redirect • Overlap or en-bloc dialing • DTMF in-band & out-of-band • Configurable call progress tones

Call Routing & Services: Regular expression number matching • Regular expression number manipulation • Least Cost Routing • Number blocking • Short-Dialing • Digit collection • Distribution- & Hunt-Groups • 2nd call offering

Voice Processing: G.711_/A-law • G.723.1 (5.3/6.3 kbps) • G.729, G.729a, G.729ab (8 kbps) • G.726 ADPCM (16 ,24, 32, 40 kbps) • G.168 echo cancellation (25ms) • Transparent ISDN data • Silence suppression and comfort noise • Adaptive and configurable dejitter buffer • Configurable packet length

Dimensions: 4.2W x 1.5H x 5.0D in. (10.6W x 3.9H x 12.7D cm) • Weight: < 15.9 oz (450 g) • Power Consumption < 4W

Environment: Temp.: 32–104°F (0–40°C) • Humidity: up to 90% , non condensing

Compliance: FCC Part 15 Class A (US EMC) • CE per RTTE 99/5/EC (EMC and LVD) • Safety - EN60950 • TBR-3 (ISDN BRI/So)

Multi-Port FXS/FXO VoIP Gateway Router

SmartNode™ 4520 Series

The SmartNode 4520 VoIP Gateway Router combines IP routing, VPN/Security, and Quality of Service for up to 8 transparent voice, fax, and data over any IP or PSTN network. Leverage low-cost packet-voice and IP services for complete branch office voice and data connectivity.

Connect with confidence using the SmartNode 4520 Series Router. Integrating a complete enterprise router with local PSTN and remote packet-voice, the SN4520 supports eight simultaneous calls for a new standard in toll-bypass, remote/branch office connectivity, and enhanced carrier services.

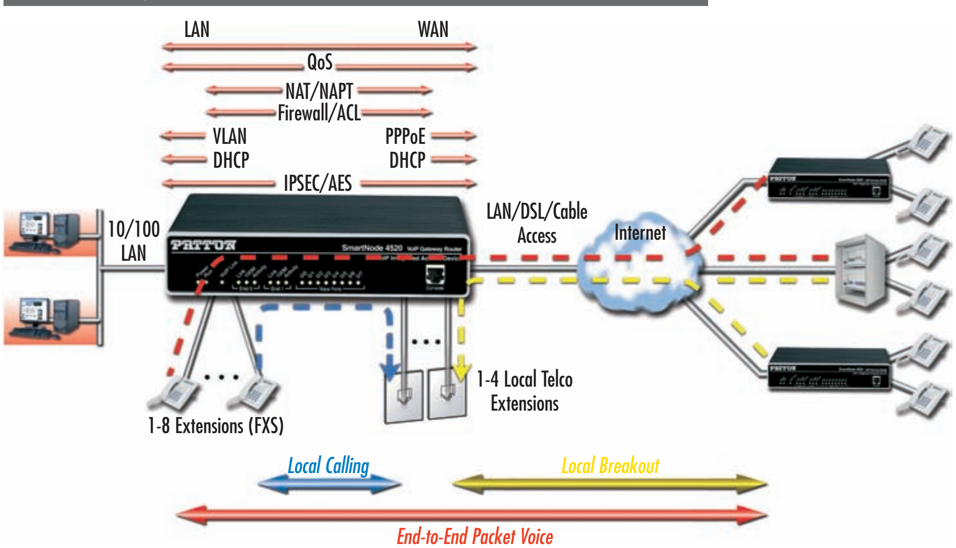
Perfect for the remote office, branch office, or PBX/switch extension, the SmartNode 4520 integrates all your voice, fax, and LAN traffic for seamless and secure networking. With its FXS analog ports the SN4520 connects to any legacy telephone or PBX and provides dial-tone, ringing, and caller-ID. When equipped with FXO ports, the local PSTN can be accessed enabling local calling and enhanced toll-bypass service.

With dual 10/100 Ethernet ports, the SN4520 provides guaranteed Quality of Service while passing LAN traffic at wire-speed. Voice traffic is prioritized while LAN/IP traffic shaping permits efficient access to the Internet and corporate networks. As a complete enterprise router, the SN4520 supports DHCP, NAT, Firewall/ACL, and PPPoE clients. While optional IPSEC VPN and VLAN features tunnel data and AES/3DES ensures secure voice over the public network.

FEATURES & BENEFITS

- ✔ Up to 8 analog ports—Compact, reliable stand-alone VoIP gateway with different port options. Supports simultaneous voice or fax calls on all ports.
- ✔ Toll-Quality VoIP—Advanced traffic management and shaping, combined with Patton's patent-pending DownStream QoS™ enforce uninterrupted toll-quality voice over best-effort networks.
- ✔ Advanced Local Call Switching—Virtual interfaces and routing tables provide industry leading flexibility in call handling programming. Local call switching, soft fallback to alternative routes. Simultaneously connects to multiple SIP services/IP PBXs.
- ✔ Complete SIP and T.38 support—Supports the complete range of industry standard VoIP: SIP, H.323, T.38 fax, fax and modem handling, DTMF relay. Codecs G.729, G.723, and so on.
- ✔ Easy Management & Provisioning—Web-based management, SNMP, command line interface. Automated mass provisioning for efficient large-scale deployments.
- ✔ Outstanding Interoperability—Proven integration for voice and T.38 fax with Asterisk™, PingTel™ and other leading IP PBX systems and soft switch vendors.

Remote Office/Branch Office Voice Extension and Access



SPECIFICATIONS

Capacity: Up to 8 simultaneous VoIP or T.38 fax calls (depending on the model)

Voice Signaling: H.323v4, SIPv2 (B2BUA capable, multi-instance, simultaneous support of multiple registrars and direct IP dialing) • SIP call transfer, redirect • DTMF in-band & out-of-band • All tones programmable (dial, ringing, busy)

Voice Processing: CODEC G.711 a-law/mu-law, G.723, G.729ab, • G.726, G.727. T.38 fax relay (9.6 k, 14.4 k) • G.711 transparent fax and bypass

Call Switching and Services: Virtual interfaces • Regular expression based call routing and number manipulation • Number blocking • Short-dialing • Digit collection, distribution and hunt groups • Transparent line extension • Fallback Routing: Soft fallback to alternative route(s)

FXS Connectivity: 2-wire Loopstart on RJ-11/12 •short haul loop 1.1km @3REN • EuroPOTS (ETSI EG201188) • programmable AC impedance, feeding, ring and on-hook voltage • Caller-ID FSK and ITU V.23/Bell 202 generation

FXO Connectivity: 2-wire Loopstart on RJ-11/12 • Programmable impedance, ring detection, tone detection, disconnect supervision • Caller ID detection

Data Services: Two 10/100 Ethernet ports • Complete IP access router • DHCP Client & server • Packet fragmentation • Static firewall, NAT, NAPT RFC 1631 access control lists • DMZ port • IPSEC, IKE, AES/DES/3DES Encryption (optional, hardware accelerated)

Quality of Service: Voice priority • DownStreamQoS™ • Traffic management, shaping and policing • IEEE 802.1p, TOS, DiffServ labeling • IEEE 802.1Q, VLAN tag insertion/deletion (simultaneous support of multiple VLANs)

Management: Web/HTTP, CLI with local console and remote Telnet access • TFTP configuration & firmware loading • SNMP MIB II and product MIB • Secure Mass provisioning for both firmware and unit/subscriber configuration • Built-in diagnostic tools (trace, debug, call generator)

System: CPU Motorola MPC875 @ 66MHz • Memory 32MB SDRAM/8MB Flash • Power 100–240 VAC (50/60 Hz) • Power dissipation 4-12W, model dependent

Temperature: 32–104°F (0–40°C)

Humidity: 5–80%, non-condensing

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 60950 • CE compliance • FCC Part 15 Class A • TBR21 (FXS) • RoHS

ORDERING INFORMATION

- SN4522/JS/EUI: 2 port FXS VoIP Gateway Router, 100–240 VAC external power supply (PS)
- SN4522/JO: 2 port FXO Gateway Router
- SN4524/JS/EUI: 4 port FXS VoIP Gateway Router
- SN4522/JO: 4 port FXO Gateway Router
- SN4526/4JS2JO: 4 port FXS 2 port FXO Gateway Router
- SN4526/JS/EUI: 6 port FXS VoIP Gateway Router
- SN4528/JS/EUI: 8 port FXS VoIP Gateway Router
- SN4524/2JS2JO/EUI: 2 port FXS, 2 Port FXO Gateway Router
- SN4528/4JS4JO/EUI: 4 port FXS, 4 Port FXO Gateway Router

Options & Accessories

- SNSW-VPN1: License Key for IPSec VPN support (DES, 3DES, AES)

Multi-Port ISDN VoIP IAD

SmartNode™ 4630 Series BRI So Gateway Router

The award-winning SmartNode 4630, with up to 5 BRI ports and 8 simultaneous voice channels, is the best way to connect ISDN networks to the world of voice over IP. It enables small offices/remote offices to lower communication costs and provides business-class Internet telephony for demanding ISDN users.

The SmartNode 4630 series are the multi-port ISDN BRI models of the proven market-leading SmartNode VoIP product family. The available 3 and 5 BRI/So port configurations fit the requirements of small and medium enterprises looking for a cost-efficient way to network PBX systems on multiple sites or connect to a public Internet telephony service.

The extra BRI port solves many VoIP network integration problems encountered in real-world installations. The port can synchronize the gateway and provide error-free ISDN data and fax transmissions, and it can be used as a fallback or local-breakout port for optimized call-routing and risk-free operation. With the life-line relay, the port even enables integration of an ISDN emergency terminal powered from the public ISDN.

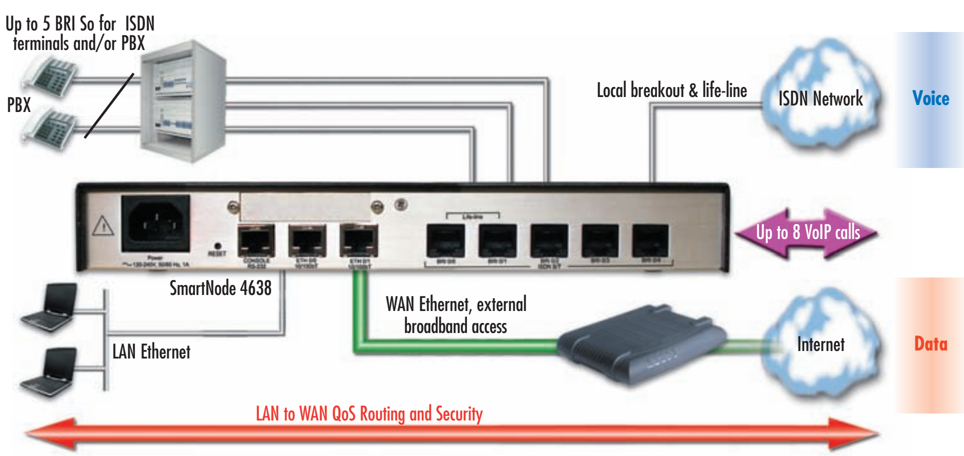
Like every SmartNode, the 4630 Series models are state-of-the-art VoIP gateways that also provide complete access routing and IP security features. Use the SmartNode as CPE or access router on broadband access, and you can benefit from industry leading Quality of Service (QoS) features ensuring a voice quality unmatched by any IP-phone or gateway on the market.

With the ClearConnect™ dial-backup option, adaptive network monitoring recognizes WAN uplink failures and initiates an ISDN dial-up connection to guarantee interrupt-free voice and data access at all times. If the VoIP link goes down or becomes congested, the SN4630 will switch over to the PSTN and guarantee your call each time.

The SmartNode 4630 is the solution for service providers and network integrators looking for a VoIP product that matches up to ISDN standards in terms of features and quality. SmartNode products provide seamless network integration, continuous trouble-free operation and cost effective deployment to protect your investments for the future.

Application—Network Integration

Whether used as a gateway or as an access router, the SmartNode 4630 provides excellent VoIP and IP QoS features for seamless network integration. All BRI ports are configurable to be TE or NT, you can thus connect your telco line(s) as well as a PBX or ISDN terminals. Terminals are powered with the built-in power supply, eliminating the need for an external box. For business class IP telephony at the tip of your fingers, the SmartNode 4630 *is more than just talk!*



FEATURES & BENEFITS

- ✔ 3/5 Ports Quality ISDN VoIP—3 or 5 ISDN BRI So ports, 4 or 8 low-bandwidth voice or T.38 fax calls. Advanced adaptive traffic management and shaping for maximum voice quality. Voice prioritization and DownStreamQoS™
- ✔ Full Telephony Features—SessionRouter™ allows flexible call routing and numbering plan adaptations, CLIP/CLIR, hold, transfer, and much more.
- ✔ Complete Access Routing—Two 10/100 Ethernet ports with auto MDI-X. Access router with NAT, Firewall, PPPoE, DHCP, DynDNS & VPN with IPsec*
- ✔ Full VoIP protocol support—SIPv2, H.323v4, MGCP/IUA, ISDN, DSS1, QSIG*, T.38, fax & modem bypass, DTMF relay.
- ✔ Interoperable for voice & T.38 fax with leading SIP service providers, soft-switch vendors, and Asterisk™ IP-PBX.
- ✔ ClearConnect™ dial-backup option for survivable voice and data connectivity.

ORDERING INFORMATION

SN4634/3BIS/UI: Multi-Port ISDN VoIP IAD—4 VoIP Call; 3 ISDN So Ports; with passthrough relay; UI power

SN4638/5BIS/UI: Multi-Port ISDN VoIP IAD - 8 VoIP Call; 5 ISDN So Ports; with passthrough relay; UI power

SW Options

SNSW-VPN1: License Key for IPsec VPN (DES, 3DES, AES), IKE and Voice-over-VPN

SNSW-QSIG1: License Key for QSIG

SNSW-DB1: Dial-Backup Feature License

Note: Please indicate country specific power cord at time of ordering.

SPECIFICATIONS

Voice Signaling: SIPv2 • H.323v4 • MGCP/IUA • SIP call transfer, redirect • Overlap or en-bloc dialing • DTMF in-band, out-of-band • Configurable tones)
Call Routing & Services: Regular expression number matching • Regular expression number manipulation • Least Cost Routing • Number blocking • Short-Dialing • Digit collection • Distribution- and Hunt- Groups
ISDN: 3/5 BRI So ports, RJ-45 • NT/TE configurable per port • Built-in line power on each port (total 4W) • DSS1, Q.921, Q.931. NTT-64 • Point-point and point-multipoint • Lifeline Bypass Relay • Optional QSIG support*
Voice Processing: G.711m/A-law • G.723.1 (6.4Kbps) • G.729, 729a, 729ab (8Kbps) • G.726 (16 ,24, 32, 40 kbps) • G.168 echo cancellation (25ms) • 4/8 simultaneous low-bandwidth voice or T.38 fax calls • Transparent ISDN data • Silence suppression and comfort noise • Adaptive and configurable dejitter buffer • Configurable packet length
IP Quality of Service: Voice priority, DownStreamQoS • Traffic Management, shaping policing • IEEE 802.1p, IEEE 802.1Q, 4096 VLANs (Tag insertion/deletion), TOS, DiffServ Labeling
Connectivity: Two 10/100Base-T Ethernet ports •Auto-MDIX • DHCP Client • PPPoE Client (multi-session) • IP Multi-Netting, VLAN, Secondary IP • IPv4, RIPv2, ICMP • Dynamic and static NAT and NAPT • ACL Firewall • DNS, DynDNS • DHCP Server • SNTP Client • Optional IPSec VPN (DES, 3DES, AES)
Management: Web-based GUI • Fully Documented CLI • Telnet and HTTP access • TFTP configuration up- and download • TFTP firmware upgrade • SNMPv1 agent (MIB II and private MIB) • Built-in diagnostic tools • Secure Auto-Provisioning
Power & Packaging: Desktop metal chassis • Dimension: 280/39/157 mm (W/H/D) • Weight: < 600g • Power Consumption < 10W
Operating Environment: Op. temp.: 32–104°F (0–40°C) Op. humidity: up to 90% , non condensing
Compliance: FCC Part 15 Class A (US EMC) • CE per RTTE 99/5/EC (EMC and LVD) • Safety–EN60950 • TBR-3 (ISDN BRI/So)

*Requires optional license.

FXS/FXO VoIP IAD with WAN Access

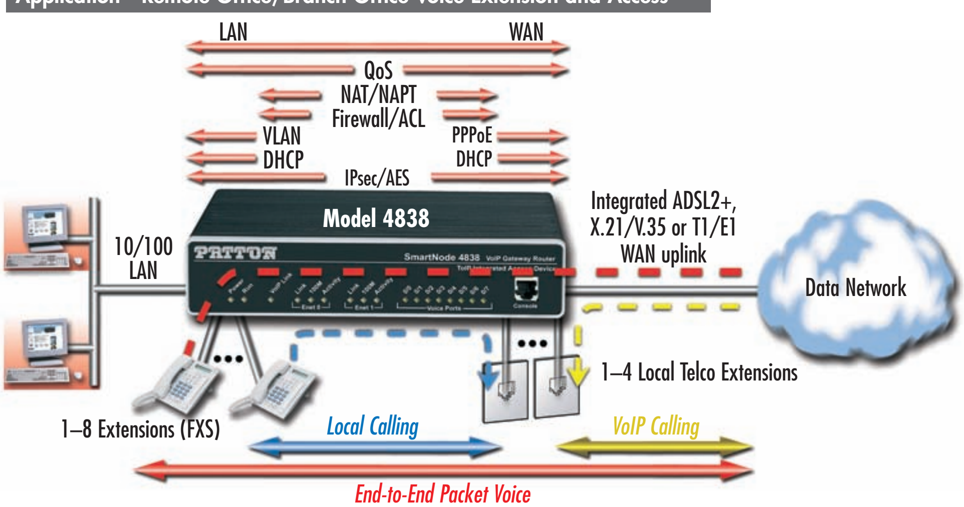
SmartNode™ 4830 Series Analog Gateway Router

The SmartNode 4830 Series is the most cost-effective VoIP IAD with integrated WAN modem in the industry. It lets you connect up to 8 phone lines with best-in-class voice quality and QoS mechanisms for voice, fax and data.

The SmartNode™ 4830 Series IADs let you deliver voice-over-IP over virtually any WAN access link type. The series offers models with combinations of 2 to 8 analog FXS phone ports and 2 or 4 FXO ports. Each model has two Ethernet ports and an integrated WAN access modem—choose from X.21, V.35, T1, E1, G.SHDSL or ADSL2+ to match the given access link with greatest flexibility. Quality of Service (QoS) features include advanced voice prioritization, traffic management, multiple VLANs and PVCs. DownStreamQoS™ ensures voice without interruptions even over best-effort internet connections. Packet labeling according to 802.1p, TOS and DiffServ enable integration into managed QoS networks.

The SN4830 is the solution for service providers and network integrators looking for the seamless integration of analog phones and PBXs into converged VoIP-Data networks. It ensures easy setup and continuous trouble-free operation and cost effective deployment. The support of the leading VoIP signaling protocols ensures interoperability with third-party equipment and protects your investments for the future.

Application—Remote Office/Branch Office Voice Extension and Access



ORDERING INFORMATION

- SN4832/JSX*/EUI: 2 Port FXS with integrated WAN
- SN4834/JSX*/EUI: 4 Port FXS with integrated WAN
- SN4836/JSX*/EUI: 6 Port FXS with integrated WAN
- SN4838/JSX*/EUI: 8 Port FXS with integrated WAN
- SN4834/2JS2JOX*/EUI: 2 FXS, 2 FXO with integrated WAN
- SN4836/ 4JS2JOX*/EUI: 4 FXS, 2 FXO with integrated WAN
- SN4832/JOX*/EUI: 2 FXO with integrated WAN
- SN4834/JOX*/EUI: 4 FXO with integrated WAN
- SN4838/4JS4JOX*/EUI: 4 FXS, 4 FXO with integrated WAN

Sync Serial Cables

- 1205-25M/35M: DB-25 male to M/34 male, for V.35 port
- 1205-25M/35F: DB-25 male to M/34 female, for V.35 port
- EMEM216006: DB-15 male to DB-15 male, for X.21 port

ADSL Splitters

- 5A-1: Single-ort ADSL Splitter

Options & Accessories

- SNSW-VPN1: License key for IPsec VPN suppport (DES, 3DES, AES)

FEATURES & BENEFITS

- ✔ Integrated WAN Access—ADSL2+, G.SHDSL, T1/E1, V.35/X.21 WAN access options. Two 10/100 Ethernet ports. Access router with NAT, Firewall, PPPoE, DHCP, and DynDNS.
- ✔ Full VoIP Protocol Support —SIPv2, H.323v4, T.38, fax & modem bypass, G.723, G.729, G.726, G.711, echo cancellation, silence compression, comfort noise, DTMF relay
- ✔ FXS, FXO, or Combinations—Up to 8 FXS ports connect to your standard telephone or PBX. 2 or 4 FXO ports allow local PSTN connections. Programmable call routing and switching.
- ✔ Maximum Voice Quality —Advanced adaptive traffic management for maximum voice quality. Voice prioritization and DownStreamQoS™.
- ✔ Management & Provisioning—Web-based management, SNMP, command line interface, & auto-provisioning for automated configuration & SW upgrades.

SPECIFICATIONS

Capacity: Up to 8 simultaneous VoIP or T.38 fax calls (depending on the model)

Voice Signaling: H.323v4, SIPv2 (B2BUA capable, multi-instance, simultaneous support of multiple registrars and direct IP dialing) • SIP call transfer, redirect • DTMF in-band & out-of-band • All tones programmable (dial, ringing, busy)

Voice Processing: CODEC G.711 a-law/mu-law, G.723, G.729ab, • G.726, G.727. T.38 fax relay (9.6 k, 14.4 k) • G.711 transparent fax and bypass

Call Switching & Services: Virtual interfaces • Regular expression based call routing and number manipulation • Number blocking • Short-dialing • Digit collection, distribution and hunt groups • Transparent line extension • Fallback Routing: Soft fallback to alternative route(s)

FXS Connectivity: 2-wire Loopstart on RJ-11/12 •short haul loop 1.1km @3REN • EuroPOTS (ETSI EG201188) • programmable AC impedance, feeding, ring and on-hook voltage • Caller-ID FSK and ITU V.23/Bell 202 generation

FXO Connectivity: 2-wire Loopstart on RJ-11/12 • Programmable impedance, ring detection, tone detection, disconnect supervision • Caller ID detection

Data Services: Two 10/100 Ethernet ports • Complete IP access router • DHCP Client & server • Packet fragmentation • Static firewall, NAT, NAPT RFC 1631 access control lists • DMZ port • IPSEC, IKE, AES/DES/3DES Encryption (optional, hardware accelerated)

Quality of Service: Voice priority • DownStreamQoS™ • Traffic management, shaping and policing • IEEE 802.1p, TOS, DiffServ labeling • IEEE 802.1Q, VLAN tag insertion/deletion (simultaneous support of multiple VLANs)

Optional WAN interfaces: X.21/V.35 Frame Relay (8 PVCs); RFC1490, FRF.12 fragmentation; LMI, Q.933D, ANSI 617D, Gang of Four; PPP, PAP, CHAP, LCP, IPCP) • T1/E1 (ITU-T G.703, ANSI T1.403; & AMI, B8ZS, HDB3), PPP • ADSL2+ (Annex A, B, I, J, I, M, U-R2) • G.SHDSL (G.991.2, Annex A, B, F, G, Up to 5.7Mbps, 8 PVCs, QoS)

Management: Web/HTTP, CLI with local console and remote Telnet access • TFTP config & firmware loading • SNMP MIB II and product MIB • Secure mass provisioning for firmware & unit/subscriber config • Built-in diagnostic tools (trace, debug, call generator)

System: CPU Motorola MPC875 @ 66MHz • Memory 32MB SDRAM/8MB Flash • Power 100–240 VAC (50/60 Hz) • Power dissipation 4–12W, model dependent

Environment: Temp.: 0–40°C • Humidity: 5–80% (non condensing)

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 60950 • CE compliance • FCC Part 15 Class A • TBR21 (FXS) • RoHS

**x* = Interface options: *C*=V.35 (DB-25F), *D*=X.21 (DB-15F), *K*=E1 (RJ-48C), *T*=T1 (RJ4-8C), *G*=G.SHDSL (RJ-11), *AYA* = ADSL Annex A, *AYB* = ADSL Annex B WAN options with ADSL 2+ annexes

Multi-Port ISDN VoIP IAD with Integrated WAN Access

SmartNode™ 4650 Series BRI So Gateway Router

The SmartNode 4650, with up to 5 BRI ports and 8 simultaneous voice channels, is the best way to connect ISDN networks to the world of voice over IP. Featuring an integrated WAN broadband modem, it delivers converged voice and data services that exceed ISDN standards.

The SmartNode™ 4650 Series are the multi-port ISDN BRI models of the proven market leading SmartNode VoIP product family. The available 3 and 5 BRI/So port configurations fit the requirements of small and medium enterprises looking for a cost-efficient way to network PBX systems on multiple sites or connect to a public Internet telephony service.

The integrated WAN access module enables you to connect the SmartNode 4650 virtually anywhere! Connected to your choice of ADSL2+, G.SHDSL, E1, T1 or V.35/X.21 broadband access, it offers complete access routing, IP security features as well as industry leading Quality of Service (QoS). The QoS features ensure a voice quality unmatched by any IP-phone or gateway on the market, and together with IGMP v2/v3, they make the SmartNode 4650 definitely 'triple-play ready'—all in a single box.

The extra BRI port solves many VoIP network integration problems encountered in real-world installations. The port can synchronize the gateway and provide error-free ISDN data and fax transmissions, and it can be used as a fallback or local-breakout port for optimized call-routing and risk-free operation. With the life-line relay, the port even enables integration of an ISDN emergency terminal powered from the public ISDN.

The SmartNode 4650 is the solution for service providers and network integrators looking for a VoIP product that matches up to ISDN standards in terms of features and quality. SmartNode products provide seamless network integration, continuous trouble-free operation and cost effective deployment to protect your investments for the future.

FEATURES & BENEFITS

- ✔ Integrated WAN Access—ADSL2+, G.SHDSL.bis, T1/E1, V.35/X.21 WAN access options. Two 10/100 Ethernet ports. Access router with NAT, Firewall, PPPoE, DHCP, & DynDNS.
- ✔ Outstanding Interoperability—Proven integration for voice and T.38 fax with leading soft switch vendors; long track record of ISDN interoperability in most countries.
- ✔ Transparent Telephony Features—Preserves ISDN features like caller ID and name (CLIP/CLIR), call transfer, hold, waiting, AOC. Supports DID and MSN lines in all major countries as well as complex numbering plan manipulations.
- ✔ Toll-Quality VoIP—Advanced traffic management and shaping, combined with Patton's patent-pending DownStream QoS™ enforce uninterrupted toll-quality voice over best-effort networks.
- ✔ Management & Provisioning—Web-based management, SNMP, command line interface. Automated provisioning for easy large-scale deployments.
- ✔ ClearConnect™ dial-backup option for survivable voice and data connectivity.—With the ClearConnect™ dial-backup option, adaptive network monitoring recognizes WAN uplink failures and initiates an ISDN dial-up connection to guarantee interrupt-free voice and data access at all times. If the link goes down or becomes congested, the SN4650 will switch over to PSTN and guarantee your call each time.

ORDERING INFORMATION

SN4654/3BISx*/UI: 3 BRI/So 4-call VoIP Router, Dual 10/100 Ethernet, Internal 90–250V power

SN4658/5BISx*/UI: 5 BRI/So 8-call VoIP Router, Dual 10/100 Ethernet, Internal 90–250V power

Sync Serial Cables

1205-25M/35M: DB-25 male to M/34 male, for V.35 port

1205-25M/35F: DB-25 male to M/34 female, for V.35 port

EMEM216006: DB-15 male to DB-15 male, for X.21 port

ADSL Splitters

5A-1: Single-port ADSL Splitter

Software options (ordered separately)

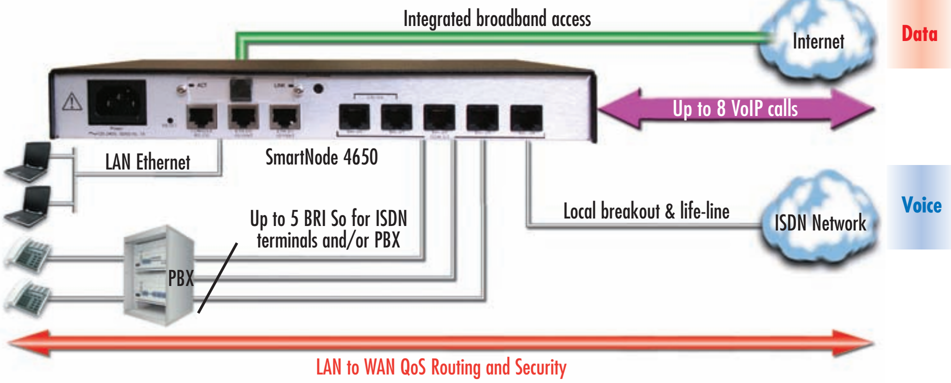
SNSW-VPN1: License Key for IPsec VPN (DES, 3DES, AES), IKE and Voice-over-VPN

SNSW-QSIG1: License Key for QSIG

SNSW-DB1: Dial-Backup Feature License

*x = Interface options: *C*=V.35 (DB-25F), *D*=X.21 (DB-15F), *K*=E1 (RJ-48C), *T*=T1 (RJ4-8C), *G*=G.SHDSL (RJ-11), *AYA* = ADSL Annex A, *AYB* = ADSL Annex B WAN options

Application diagram



SPECIFICATIONS

Capacity: 4 (SN4654 models) or 8 (SN4658 models) simultaneous VoIP or T.38 fax calls

ISDN Connectivity: • 5 ports Euro-ISDN BRI/So RJ-45 • All configurable TE or NT side • Built-in line power • DSS-1, Q.921, Q.931 • Point-point and point-multipoint • Lifeline Bypass Relay • Optional QSIG support

VoIP Signaling: SIPv2 (B2BUA capable, multi-instance, supports registrar and direct IP dialing at the same time) • H.323v4 • MGCP/IUA • SIP call transfer, redirect • DTMF in-band & out-of-band • All tones programmable (dial, ringing, busy) • Overlap or en-bloc dialing• Transparent AOC, ECT, CLIP, CLIR, etc • speech, audio & data (Fax Gr 4, UDI 64, • RDI 64);

Voice Processing: CODEC G.711 a-law/mu-law, G.723, G.729ab, • G.726, G.727. T.38 fax relay (9.6 k, 14.4 k) • G.711 transparent fax and bypass

Management: Web/HTTP, CLI with local console and remote Telnet access • TFTP configuration & firmware loading • SNMP MIB II & product MIB • Secure mass provisioning for both firmware and unit/subscriber configuration • Built-in diag tools (trace, debug, call generator)

Data Services: Two 10/100 Ethernet ports • Complete IP access router • DHCP Client & server • Packet fragmentation • Static firewall, NAT, NAPT RFC 1631 access control lists • DMZ port • IPSEC, IKE, AES/DES/3DES Encryption (optional, hardware accelerated)

Quality of Service: Voice priority • DownStreamQoS™ • Traffic management, shaping and policing • IEEE 802.1p, TOS, DiffServ labeling • IEEE 802.1Q, VLAN tag insertion/deletion (simultaneous support of multiple VLANs and PPPoE sessions)

Optional WAN interfaces: • X.21/V.35 Frame Relay (8 PVCs); RFC1490, FRF.12 fragmentation; LMI, Q.933D, ANSI 617D, Gang of Four; PPP, PAP, CHAP, LCP, IPCP) • T1/E1 (ITU-T G.703, ANSI T1.403; & AMI, B8ZS, HDB3), PPP • ADSL2+ (Annex A, B, I, J, I, M, U-R2) • G.SHDSL (G.991.2, Annex A, B, F, G, Up to 5.7Mbps, 8 PVCs, QoS)

System: CPU Motorola MPC870 @ 66MHz • Memory 32MB SDRAM/8MB Flash • Power 100–240 VAC (50/60 Hz) • Power dissipation 4-12W, model dependent

Environment: Temp.: 0–40°C • Humidity: 5–80% (non condensing)

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 60950 • CE compliance • FCC Part 15 Class A • TBR3 (ISDN) • RoHS

Multi-Port T1/E1 VoIP Integrated Access Device

SmartNode™ 4960

The award-winning SmartNode 4960 integrates with legacy telephony gear to deliver VoIP and data services with QoS and encrypted-voice VPNs. The SmartNode 4960 comes with four T1/E1/PRI ports, two GigE ports, and supports up to 120 simultaneous VoIP calls, making it the ideal choice for low-cost, secure, prioritized communications.

Providing a high-density seamless link between the circuit-switched telephone network and voice-over-IP, the SN4960 is ideal for PBX business trunking or corporate VoIP access. Offering up to four software configurable T1/E1/PRI interfaces the SN4960 connects to any switch, PBX and data network with up to 120 simultaneous calls using SIP, T1, E1 or PRI signaling. The dual gigabit Ethernet ports connect to the network for the highest throughput with its integrated QoS router. With its built-in CSU/DSU, any T1/E1 port can be selected as a WAN port for a truly integrated voice and data access

Like every SmartNode, the SN4960 delivers toll-quality voice with all industry standard CODECs including low-bandwidth G.723/G.729. Business class services are supported with T.38 fax, fax bypass and modem bypass features.

With the ClearConnect™ dial-backup option, adaptive network monitoring recognizes WAN uplink failures and initiates an ISDN dial-up connection to guarantee interrupt-free voice and data access at all times. If the VoIP link goes down or becomes congested, the SN4960 will switch over to the PSTN and guarantee your call each time.

The SmartNode 4960 is ready for SIP TLS and SRTP through software upgrades. Exclusive DownStreamQoS™ and Voice-over-VPN features give the clear advantage of uninterrupted and secure voice communication for any call today.

SPECIFICATIONS

Voice Connectivity: Up to four software selectable T1/E1/PRI ports • Signalling support (ISDN DSS-1, NI-2, Q.SIG; CAS Robbed bit loop and ground start, E&M, immediate, wink, double wink) • SIPv2 & MGCP/IUA, H.323v4 • ISDN AOC/ECT • ISDN speech, audio & data (Fax Gr 4, UDI 64, • RDI 64); ISDN supplementary services

Voice processing: Codec G.711 a-law/mu-law, G.723, G.729ab, • G.726, G.727. T.38 fax relay (9.6 k, 14.4 k) • G.711 transparent fax and bypass

Call routing and services: Regular expression matching and manipulation; number blocking; short-dialing; digit collection, distribution and hunt groups.

Data interfaces: Dual 10/100/1000 TX Ethernet Ports • Autosensing • Auto-MDI • Full-duplex

IP Routing: Complete IP access router • DHCP Client & server • Packet fragmentation • Static firewall, NAT, NAPT RFC 1631 access control lists

IP Quality of Service: Voice priority, DownStreamQoS • traffic management, shaping policing • IEEE 802.1p, TOS, DiffServ labeling • IEEE 802.1Q, VLAN tag insertion/deletion 4,096

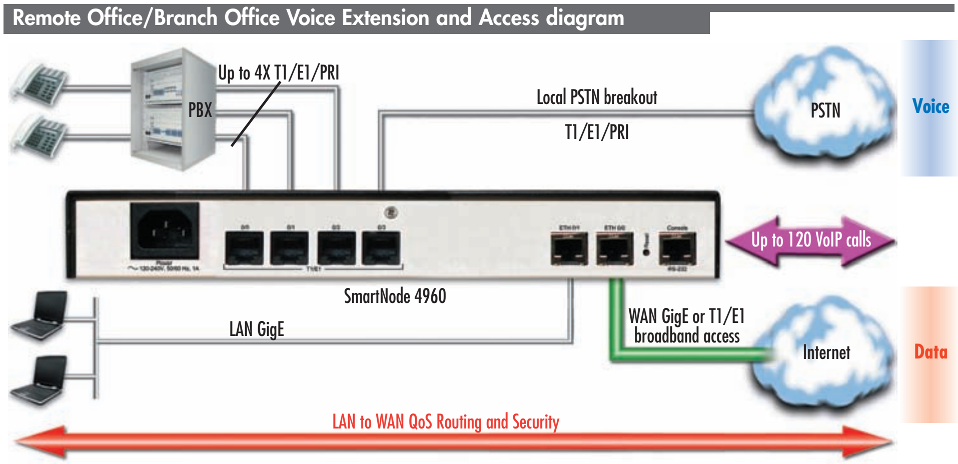
Management: Web/HTTP, CLI with local console & remote Telnet access • TFTP configuration & firmware loading • SNMP MIB II and product MIB • Secure autoprovisioning for firmware & unit/subscriber configuration • Built-in diagnostic tools (trace, debug, call generator)

Environment: Temp: 32–104°F (0–40°C); Humidity: Up to 90% (non condensing)

Power: 100–240 VAC (50/60 Hz) Power consumption: 15W

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 60950 • CE compliance • FCC Part 15 Class A; Part 68; CS-03

Remote Office/Branch Office Voice Extension and Access diagram



FEATURES & BENEFITS

- ✔ Up to 120 simultaneous voice or T.38 fax calls with one to four T1/E1/PRI ports and dual Gigabit Ethernet ports. Use any CODEC or fax on any port, any time.
- ✔ Universal SIP and T.38 support—Softswitch certified signaling support between all T1 RBS CAS, ISDN PRI, Q.SIG, SIP, H.323 and MGCP/IUA protocols.
- ✔ Secure Toll-Quality VoIP—DownStreamQoS and Voice-over-VPN with adaptive traffic management and shaping for maximum voice quality and secure voice communication.
- ✔ Transparent Telephony Features—Handles complex number manipulation and mapping scenarios for most seamless integration with existing infrastructure, CLIP, CLIR, hold, transfer and much more.
- ✔ Management & Provisioning—Web-based management, SNMP, command line interface. Automated provisioning for easy large-scale deployments.
- ✔ ClearConnect™ dial-backup option for survivable voice and data connectivity.

ORDERING INFORMATION

SN4960/1E15V/UI: SmartNode Hi-Cap 1 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 15 VoIP Channels, upgradeable to 30 calls.

SN4960/1E24V/UI: SmartNode Hi-Cap 1 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 24 VoIP Channels, upgradeable to 30 calls.

SN4960/1E30V/UI: SmartNode Hi-Cap 1 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 30 VoIP Channels, non-upgradeable.

SN4960/4E15V/UI: SmartNode Hi-Cap 4 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 15 VoIP channels, field upgradeable to a max of 60 channels.

SN4960/4E24V/UI: SmartNode Hi-Cap 4 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 24 VoIP channels, field upgradeable to a max of 60 channels.

SN4960/4E30V/UI: SmartNode Hi-Cap 4 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 30 VoIP channels, field upgradeable to a max of 60 channels.

SN4960/4E48V/UI: SmartNode Hi-Cap 4 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 48 VoIP channels, field upgradeable to a max of 60 channels.

SN4960/4E60V/UI: SmartNode Hi-Cap 4 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 60 VoIP channels, non-upgradeable

SN4960/4E96V/UI: SmartNode Hi-Cap 4 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 96 VoIP channels, field upgradeable to a max of 120 channels.

SN4960/4E120V/UI: SmartNode Hi-Cap 4 T1/E1/PRI VoIP IAD, 2x GigEthernet, UI power, 120 VoIP channels.

Options & Accessories

SNSW-49V6: 6 channel Voice Upgrade Key for SN4960 VoIP IADs. Software expansion for additional voice channels.

SNSW-VPN2: Software option for IPsec VPN, including DES/3DES and AES encryption, IKE and Voice-Over-VPN.

SNSW-QSIG2: Support for ISDN Q.SIG.

SNSW-DB2: Dial-Backup Feature License

IpChannel Bank Multi-Port FXS & FXO Gateway Router

SmartNode™ 4900 Series

The IpChannel Bank is the perfect VoIP gateway for applications requiring 12 to 32 concurrent analog voice/fax calls within a single redundant solution.

The SmartNode 4900 is the ideal solution for service providers and enterprises requiring high-density analog connections for converged Internet-Telephony. Call centers, multi-tenant-units and PBX/switch extensions can now access the low-cost benefits of packet voice while WAN, data and VPN features permit direct access the IP network with full upstream and DownStream QoS™.

The SN4900 supports 12 to 32 simultaneous VoIP calls over standard two-wire FXS connections. The analog ports are presented on a single Amphenol telco connector for convenient wiring closet connection. Local LAN connectivity is presented via dual 10/100 Ethernet ports. Traffic can be routed out either port for load-balancing and redundancy. Dual-redundant power supplies protect against equipment down-time.

Seamless and secure network integration with fixed IP, DHCP or PPPoE. Complete access routing features include NAT/NAPT, Firewall, DynDNS and the optional IPSec VPN feature license offer secure data.

Optional WAN uplink modules, available in V.35/X.21, T1/E1, and xDSL options, eliminate the need for extra network termination devices and extra cost.

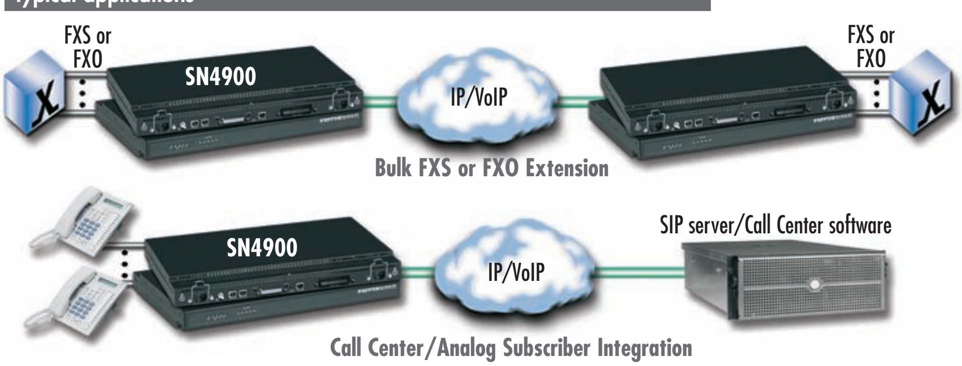
Quality of Service (QoS) features included advanced voice prioritization and traffic management. DownStreamQoS™ ensures voice without interruptions even over best-effort internet connections. Packet labeling according to 802.1p, TOS and DiffServ enable integration into managed QoS networks.

Integrated GUI and Command Line, front panel status and call load indicators and a full suite of management interfaces ensure efficient setup, continuous trouble-free operation and cost-effective deployment.

FEATURES & BENEFITS

- ✔ 12, 16, 24 or 32 FXS or FXO ports—Simultaneous voice or fax calls on all ports. Advanced local call switching.
- ✔ Full SIP and T.38 support—Supports the complete range of industry standard VoIP: SIP, H.323, T.38 fax, fax and modem bypass, DTMF relay. Codecs G.729, G.723 etc.
- ✔ Secure Toll-Quality VoIP—DownStreamQoS and Voice-over-VPN with adaptive traffic management and shaping for maximum voice quality and secure voice communication.
- ✔ Complete Access Routing—Two 10/100 Ethernet ports with auto MDI-X. Access router with NAT, Firewall, PPPoE, DHCP, DynDNS, multiple VLANs & VPN with IPSec*.
- ✔ Optional Integrated WAN uplink—Choose from V.35, X.21, T1/E1, ADSL and G.SHDSL data interfaces in addition to the two Ethernet ports.
- ✔ Outstanding Interoperability—Interoperable for voice and T.38 fax with leading SIP service providers, softswitch vendors and Asterisk™ IP-PBX.
- ✔ Use FXS models for call center applications and FXO for PSTN analog trunking.

Typical applications



ORDERING INFORMATION

- SN4912/JS/RUI: IpChannel Bank 12 Port FXS
- SN4916/JS/RUI: IpChannel Bank 16 Port FXS
- SN4924/JS/RUI: IpChannel Bank 24 Port FXS
- SN4932/JS/RUI: IpChannel Bank 32 Port FXS
- SN4912/JSX*/RUI: IpChannel Bank 12 Port FXS WAN Uplink
- SN4916/JSX*/RUI: IpChannel Bank 16 Port FXS WAN Uplink
- SN4924/JSX*/RUI: IpChannel Bank 24 Port FXS WAN Uplink
- SN4932/JSX*/RUI: IpChannel Bank 32 Port FXS WAN Uplink
- SN4912/JO/RUI: IpChannel Bank 12 Port FXO
- SN4916/JO/RUI: IpChannel Bank 16 Port FXO
- SN4924/JO/RUI: IpChannel Bank 24 Port FXO
- SN4932/JO/RUI: IpChannel Bank 32 Port FXO
- SN4912/JOX*/RUI: IpChannel Bank 12 Port FXO, WAN uplink
- SN4916/JOX*/RUI: IpChannel Bank 16 Port FXO, WAN uplink
- SN4924/JOX*/RUI: IpChannel Bank 24 Port FXO, WAN uplink
- SN4932/JOX*/RUI: IpChannel Bank 32 Port FXO, WAN uplink

Options & Accessories

SNSW-VPN1: License Key for IPSec VPN support (DES, 3DES, AES)

Note: 48VDC or split 48VDC/UI power options available.

*X = Interface options: C=V.35, D=X.21, K=E1, T=T1, Fi=Fiber, AYx=ADSL, G=G.SHDSL

SPECIFICATIONS

Capacity: 12, 16, 24, 32 simultaneous VoIP calls

Voice Signaling: SIPv2 H.323v4 (simultaneously with B2BUA capability) • SIP call transfer, redirect • DTMF in-band & out-of-band • All tones programmable (dial, ringing, busy)

Voice Processing: CODEC G.711 a-law/mu-law, G.723, G.729ab, • G.726, G.727. T.38 fax relay (9.6 k, 14.4 k) • G.711 transparent fax and bypass

Call Switching and Services: Regular expression based call routing and number manipulation • Number blocking • Short-dialing • Digit collection, distribution and hunt groups • Transparent line extension

FXS Connectivity: 2-wire Loopstart on 50pin (12 to 24 channels) or 64pin (32 channels) Telco connector •short haul loop 1.1km @3REN • EuroPOTS (ETSI EG201188) • programmable AC impedance, feeding, ring and on-hook voltage • Caller-ID FSK and ITU V.23/Bell 202 generation

FXO Connectivity: 2-wire Loopstart on 50pin (12 to 24 channels) or 64pin (32 channels) Telco connector • Programmable impedance, ring detection, tone detection, disconnect supervision • Caller ID detection

Data Services: Two 10/100 Ethernet ports • Complete IP access router • DHCP Client & server • Packet fragmentation • Static firewall, NAT, NAPT RFC 1631 access control lists • DMZ port

Quality of Service: Voice priority • DownStreamQoS™ • Traffic management, shaping and policing • IEEE 802.1p, TOS, DiffServ labeling • IEEE 802.1Q, VLAN tag insertion/deletion 4,096

Optional WAN interfaces: X.21/V.35 Frame Relay (8 PVCs); RFC1490, FRF.12 fragmentation; LMI, Q.933D, ANSI 617D, Gang of Four; PPP, PAP, CHAP, LCP, IPCP) • T1/E1 (ITU-T G.703, ANSI T1.403; & AMI, B8ZS, HDB3) • ADSL2+ (Annex A, B, I, J, I, M, U-R2) • G.SHDSL (G.991.2, Annex A, B, F, G, Up to 5.7Mbps, 8 PVCs, QoS)

Management: Web/HTTP, CLI with local console and remote Telnet access • TFTP configuration & firmware loading • SNMP MIB II and product MIB • Secure auto-provisioning for both firmware and unit/subscriber configuration • Built-in diagnostic tools (trace, debug, call generator)

System: CPU Motorola MPC875 @ 133 MHz • Memory 32MB SDRAM/8MB Flash

Power: 100–240 VAC (50/60 Hz) • Power dissipation: > 22W (60W max, model SN4932/JS/RUI)

Environment: Temp.: 0–40°C • Humidity: 5–80% (non condensing)

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 50950 • CE compliance • FCC Part 15 Class A

4-Slot Modular Router

SmartNode™ 2400

Combining a real-time QoS IP router, Voice-over-Packet Gateway, and ToIP circuit switching, the SmartNode 2400 Series VoIP Media Routers & Gateways are ideal for multi-service carrier access and corporate PSTN networking for up to 120 simultaneous calls.

The SmartNode 2400 Series provides a high density, seamless link between the circuit-switched telephone network and Voice-over-IP. Ideal for multi-service carrier and corporate PSTN access, the SmartNode 2400 Series takes 120 calls from voice-to-packet. With dual 10/100 Ethernet ports and four voice interface slots, the SmartNode 2400 connects to any switch, PBX, or data network.

The SmartNode delivers VoIP that has the clarity of toll-quality voice. Multiple user-defined G.711 and G.726 voice CODECs are included, as well as G.723 and G.729ab for low bandwidth voice. T.38 FoIP, fax bypass, and modem bypass capabilities ensure that the SmartNode will seamlessly connect to all voice and data services. QoS ensures optimal voice performance through traffic classification and prioritization.

Patton's exclusive Telephony-over-IP™ (ToIP™) programmable circuit switching delivers service transparency and flexible PSTN integration. With ToIP any BRI, T1/E1 PRI, or FXS port can connect to any telephone or media gateway port.

FEATURES & BENEFITS

- ✔ Supports up to 120 simultaneous calls from any combination of voice signaling including T1 RBS, T1/E1 PRI, BRI, and FXS.
- ✔ T1/E1, BRI, & FXS Voice Interface Cards—Use any combination of voice interfaces for connection to any telephone, switch, or service provider.
- ✔ Expansion slots for ISDN, T1, E1, and FXS interface cards for voice processing features
- ✔ Voice VPNsReal-time quality of service (TOS, Diffserv, and IEEE 802.1p/Q)
- ✔ Simultaneously compresses up to 120 ISDN voice calls over the same IP link

ORDERING INFORMATION

Note: Select up to four ISDN, T1, E1, and FXS interface cards from page 31 to configure the SmartNode Gateway Router to meet your requirements.

SN2400/OVIL/UI: Gateway Router QoS ToIP, Internal UI (90–260 VAC) power; no VoIP modules

Options & Accessories

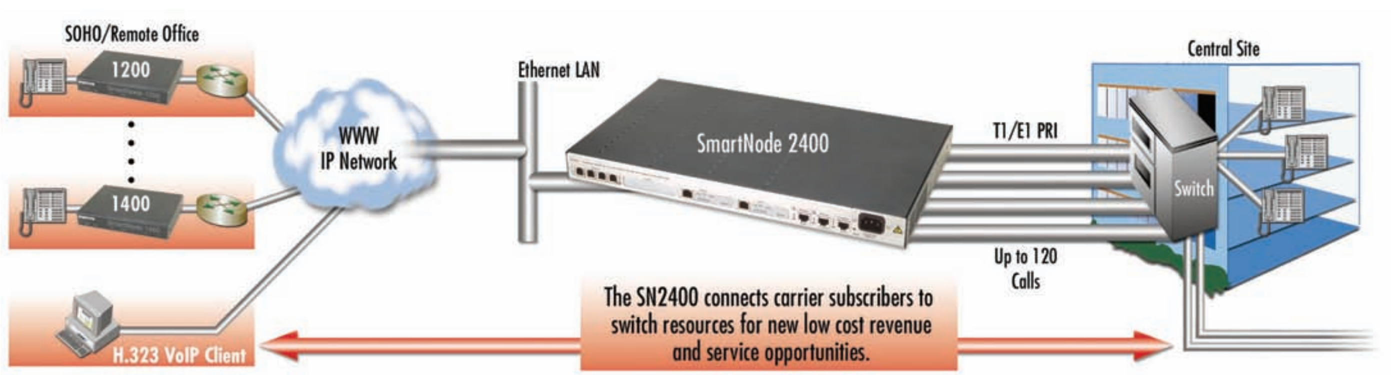
SNSW-VPN2: License Key for IPSec VPN support (DES, 3DES, AES)

Application—Shared-Switch Access

To take advantage of emerging technologies and the changing regulatory environments, network service providers are providing new services which can be integrated within the enterprise to reduce costs and improve service. Deployed as a trunk gateway, the SmartNode is the seamless link between the PSTN and VoIP access. Subscribers now access new carrier services without incurring charges from the incumbent carrier.

In enterprise networks, the SmartNode integrates telephony and IP data communications for best use of bandwidth, improved office-to-office communication, and reduced network costs.

Instead of installing a separate PBX in a remote office, the SmartNode is able to provide transparent extension of PBX phones. The extension can be managed centrally and benefit from services such as calling groups, least cost routing, and call forwarding.



SPECIFICATIONS

Data Connectivity: 2 10/100Base-T Ethernet, RJ-45

Voice Signalling: Euro ISDN EDSS-1/ETSI BRI/NET3 (ETS 300 012-1 (ITU-T I.430); ETS 300 402-1 (ITU-T Q.921); ETS 300 403-1/2 (ITU-T Q.931); ETS 300 102-1 (ITU-T Q.931)) • Q-SIG (PSS-1) (ECMA-143; ETSI and ISO/ECMA channel numbering) • SIP and MGCP • H.323v3 (RAS, H.225, H.245; Fast-connect, early H.245; Gatekeeper autodiscovery; Alias registration; Overlap sending; Empty capability set (call transfer, hold); H.323v1 call transfer, hold) • ECS support • ISDN over IP (ISoIP) • H.323 GW and GK compatible • H.323 Annex M3 • ISDN/Q-SIG feature tunnelling • ISDN speech, audio and data (Fax Gr 4, UDI 64, RDI 64) • ISDN supplementary services

Voice Routing—Session Router™: Local switching • Interface huntgroups • ISDN broadcast message routing • Routing Criteria (Interface; Calling/called party number; Time of day, day of week, date; ISDN bearer capability) • Number manipulation functions (Replace numbers; Add/remove digits) • Multiple remote gateways • PLAR

IP Routing: IPv4 router • RIPv1, v2 (RFC 1058 and 2453) • Programmable static routes • ICMP redirect (RFC 792) • DHCP client/server • Packet fragmentation • Static Firewall (NAT/PAT/NAPT (RFC 1631); Access control lists)

IP Quality of Services: WFQ/Fixed Rate/Priority Queuing/Flow-split scheduler • Combination of QoS schemes with configurable burst • DiffServe/ToS set or queue per header bits • Packet Policing discards excess traffic • 802.1p/Q VLAN support with 4096 IDs • Traffic classification

Management: Industry standard CLI • Local console (RS-232, RJ-45) and remote Telnet access • TFTP configuration down- and upload • TFTP firmware download • SNMP v1 agent (MIB II and private MIB) • Built-in diagnostic tools (trace, debug) • Java™ Applet • HPOV Integration with NNM

Operating Environment: Op. temperature: 0–40°C
Op. humidity: 5–80% (non-condensing)

System: CPU Motorola MPC750 at 333 MHz (Memory 32MB SDRAM (160MB max.)/16 MB Flash) • Power: 100–240 VAC (50/60 Hz) • Power dissipation 30W (fully loaded)

Compliance: EMC compliance: EN55022 and EN55024 • Safety compliance: EN 50950 • CE compliance • FCC Part 15 Class A

30-Channel E1 Module for ISDN PRI

IC-E1V Interface Card for SmartNode™ 2400

ORDERING INFORMATION

SN-IC-E1V: 30-channel E1 gateway interface card for ISDN PRI
SN-IC-E1V-0: E1 Gateway interface card for ISDN PRI (circuit switching only)
SN-IC-E1V-15: 15-channel E1 gateway interface card for ISDN PRI

FEATURES & BENEFITS

- ✔ IC-E1V provides one PRI E1 port to support 30 VoIP calls. For mounting in the SN2400 expansion slots.
- ✔ IC-E1V-0 is an E1 gateway interface card for ISDN PRI-circuit switching only.
- ✔ IC-E1V-15 is a 15-channel gateway interface card with one ISDN PRI/S2m port.

23-Channel T1 Module for ISDN PRI

IC-T1V Interface Card for SmartNode™ 2400

ORDERING INFORMATION

SN-IC-T1V: 23-channel T1 gateway interface card for ISDN PRI
SN-IC-T1V-0: T1 Gateway interface card for ISDN PRI (circuit switching only)
SN-IC-T1V-15: 15-channel T1 gateway interface card for ISDN PRI

FEATURES & BENEFITS

- ✔ IC-T1V provides one PRI T1 port to support 30 VoIP calls. For mounting in the SN2400 expansion slots.
- ✔ IC-T1V-0 is an T1 gateway interface card for ISDN PRI-circuit switching only.
- ✔ IC-T1V-15 is a 15-channel gateway interface card with one ISDN PRI/S2m port.

8-Channel Module Provides 4 ISDN BRI Interfaces

IC-4BRV Interface Card for SmartNode™ 2400

ORDERING INFORMATION

SN-IC4BRV-8V: 4 x ISDN BRI/So VoIP, NT/TE configurable, 8 DSP VoIP channels
SN-IC4BRV-8VR: 4 x ISDN BRI/So VoIP, NT/TE configurable, 8 DSP VoIP channels

FEATURES & BENEFITS

- ✔ IC-4BRV provides four BRI/S0 ISDN ports to support 8 VoIP calls. For mounting in the SN2400 expansion slots.
- ✔ IC-4BRV-8V is an 8-channel gateway interface card for ISDN BRI/S0.
- ✔ IC-4BRV-8VR is an 8-channel gateway interface card for ISDN BRI/S0 with hardware bypass (emergency) relay.

4-Channel Module Provides 4 Analog Phone Ports

IC-4FXS Interface Card for SmartNode™ 2400

ORDERING INFORMATION

SN-IC4FXS: 4 x FXS VoIP, 4 DSP VoIP channels Requires PM-4XV-int for FXS power

FEATURES & BENEFITS

- ✔ IC-4FXS provides 4 analog FXS ports and connects to the SmartNode base unit through a PCI packet and PCM circuit interface. The IC-4FXS supports up to 4 simultaneous voice or fax calls.

I'm Natalie, Patton's Inside Sales Manager, US & Canada. Call me at ***+1 301.975.1000*** *when you want to purchase Patton products or if you have questions about our products. You can also send e-mail to* ***sales@patton.com****.*

IP-PBX

SipXNano-15 & SipXNano-30

Patton's SipXNano IP-PBX enables small enterprises to access large enterprise PBX features and becomes the first to realize the benefits of VoIP.

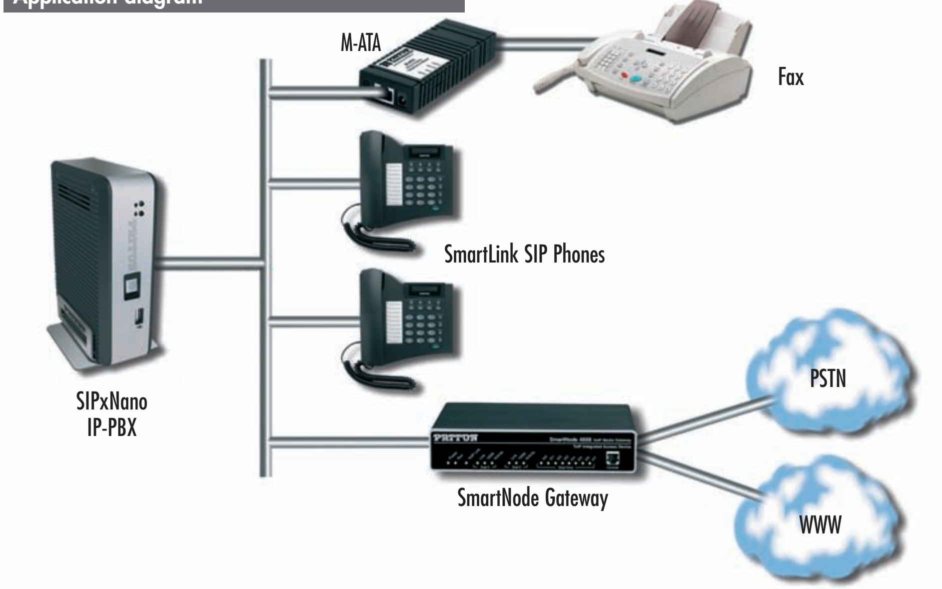
The SIPxNano is a full function IP-PBX tailored to meet the needs of small to medium enterprises and branch offices. It provides the features of a traditional PBX with the benefits of VoIP such as soft clients and voicemail to e-mail integration. The SIPxNano supports up to 30 concurrent users and comes with common PBX/Key System features such as an auto-attendant, configurable call routing, music on hold, call-forwarding, music on park, hunt groups, and much more.

The web-based administrative interface permits quick and easy configuration of dialing plans, auto attendants and other PBX features. Office users of the SIPxNano IP-PBX benefit the most from the voicemail to e-mail integration. An easy to use web page is available to create and manage personal voicemail folders. Menu options facilitate the setup of e-mail notification and even e-mail delivery of the message in easy-to-open .wav file format. Multiple greetings can be recorded and then activated via a drop drown box.

Enterprises operating in multiple locations can leverage their existing data intranet/VPN investment to extend features of the headquarters IP-PBX to all locations and telecommuters. Every employee can be reached via a 3-digit extension by a simple web page configuration of the dialing plan. Intra-enterprise long-distance phones bills drop to zero by using the VPN. At the same time, additional savings can be achieved both by reducing the number of local PSTN connections, as well as by leveraging the remaining local PSTN connections in remote offices to effect a least cost routing of long distance calls to the remote office area code and thereby performing a long distance toll by-pass.

When combined with a Patton SmartNode or SmartLink device, the SIPx-Nano IP-PBX provides VoIP services to analog phones, faxes, and legacy PBX systems that are not able to support newer VoIP technologies.

Application diagram



FEATURES & BENEFITS

- ✔ Full Function Enterprise PBX—Complete call control, voicemail and administrative systems
- ✔ Drives Intra-Enterprise Calling Costs Down—All voice traffic is transported as data across your data connection on the enterprise IP network
- ✔ Geographically Unified Calling Features —All employees with an Internet connection can have access to the same PBX features from any location
- ✔ Easy to Use System Configuration—Full system and user administration via web interface
- ✔ Voicemail and Email Integration—Sends your voicemail to your email
- ✔ Web-Based Self-Administration—Users can specify call forwarding, call routing and voicemail preferences thru the web interface

SPECIFICATIONS*

Protocol: IETF SIP RFC3261—SIP

Call Routing/Call Processing: Automatic call route selection • User and Administrator web-based configuration • Call admission control • Digitmap-based call setup control • Dynamic call forwarding • Hotline/Ring down • Hunt groups • Message waiting indication • Multi-site/Multi-location • Multi-station appearance • Local/Remote stations • Outbound Call blocking/Toll restriction • Out-of-band DTMF signaling • Phone number to SIP address alias facility • System security • Web services APIs for configuration server • User forwarding through GUI • Backup configuration via the GUI • Emergency routing • Attendant console • Direct call pick-up • Call Park/Retrieve with music on hold

Media Server: HTTPS-based message storage • Customizable voicemail greetings • Operator (zero out) • Login to VM via user greeting • User distribution lists • Change VM password through GUI

Auto Attendant: Alpha-dialed number confirmation • Customizable main greeting • Dial by extension • Dial by Name • Operator (zero out)

Interactive Voice Response (IVR): Customizable top-level IVR choices

CPU : Onboard VIA EDEN N Nano processor (1.3 GHz Pentium equivalent)

RAM: 256Mb DDR266

I/O Interface: LPT • RS-232 • PS/2 Keyboard • PS/2 Mouse • USB 2.0 ports, Qty: 2

Operating System: Linux

Power supply: Universal 100–240 VAC

Dimensions: 6.7 x 4.9 x 2.3 in. (170 x 124 x 58 mm) Aluminum case

Weight: 940g

Operating Humidity: 0–90% relative humidity, non-condensing

Operating Temp.: 32–140°F (0–60°C)

* Hardware specifications subject to change without notice.

ORDERING INFORMATION

SIPxNano-15/E: SmartLink IP- PBX NanoServer & 15 concurrent user license with 1-year software maintenance; AC UI power

SIPxNano-30/E: SmartLink IP-PBX NanoServer & 30 concurrent user license with 1-year software maintenance; AC UI power

SL4250-75: SmartLink IP-PBX rack mount server w/RAID1 & 75 user license with 1-year software maintenance

SL4250-125: SmartLink IP-PBX rack mount server w/RAID1 & 125 user license with 1-year software maintenance

SL4250-250: SmartLink IP-PBX rack mount server w/RAID1 & 250 user license with 1-year software maintenance

SIP Telephones

SmartLink™ 4050 Series

Move to the world of VoIP telephony with these full-function SIP phones.

SL4050/2

Smartlink 4050 IP Phones are full-featured VoIP telephones that provide a rich set of end user calling features and allow full integration with intranet IP-PBXs and Internet Telephony Service providers. With speaker phone, multi-line support, one touch dialing, local conferencing capabilities, and 200 number local phone directory, the SL4050 phones are suitable for the corporate desktop or the home/home office.

Full support is provided for local configuration of call handling for busy, call forward and conditional call forward.

The SmartLink 4050s can be easily installed and configured through a local web interface for individual installations. For enterprises or service providers in need of large scale deployments, the SIP phones feature automatic firmware and configuration and phone directory download from a central configuration server for plug-and-play setup.

FEATURES & BENEFITS

- ✔ Full VoIP SIP feature set
- ✔ Rich set of local calling features including conditional call forwarding, call park, and group pickup
- ✔ Dual Ethernet ports to minimize cabling needs
- ✔ Up to four SIP user accounts
- ✔ Up to 10 lines or one-touch speed dial buttons
- ✔ Local Web configuration or remote configuration download
- ✔ Local conferencing
- ✔ Call transfer
- ✔ Message waiting indicator

ORDERING INFORMATION

SL4050/2/EUI/K: 2 Line VoIP SIP Phone with 2X 10/100Base-TX, EUI power, North American power cord. Color: Black

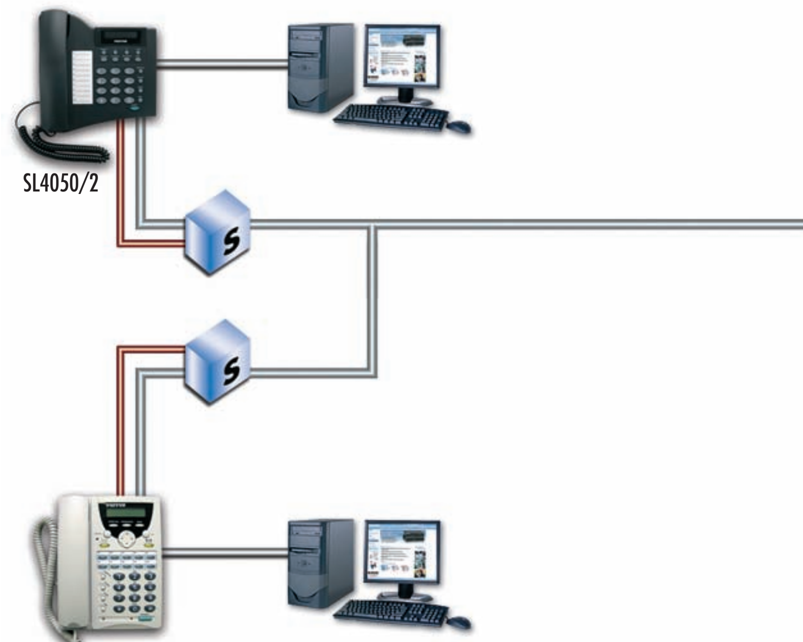
SN4050/2/EUI/A: 2 Line VoIP SIP Phone with 2X 10/100Base-TX, EUI power, European power cord. Color: Black

SL4050/2/E: 2 Line VoIP SIP Phone with 2X 10/100Base-TX, EUI power, Order cord or PoE independently power cord. Color: Black

SL4050/10/EUI/K: 10 Line VoIP SIP Phone with 2X 10/100Base-TX, EUI power, North American power cord. Color: White

SL4050/10/EUI/A: 10 Line VoIP SIP Phone with 2X 10/100Base-TX, EUI power, European power cord. Color: White

SL4050/10/E: 10 Line VoIP SIP Phone with 2X 10/100Base-TX, EUI power, Order cord or PoE independently power cord Color: White



Options & Accessories for SN4050/2 & SN4050/10

P0805: EUI 110–240V to 5V power transformer—Order country-specific power cord

0805-INF-POE-5: IEEE 802.3af Power over Ethernet (PoE) splitter—5VDC output

SPECIFICATIONS

Protocol: IETF SIP RFC3261

Network Interface: RJ45 x 2, 10/100Base-T

Call Features: Call transfer (unattended/blind & announced) • Call forward (busy/no answer/unconditional) • anonymous call blocking • out-of-band DTMF (RFC 2833) • message waiting indicator • call park/pickup (support SIP required) • group pickup (Support SIP server required)

Voice Codec: G.711µ-law • G711a-law • G.723.1 (5.3k) • G.723.1 (6.3k) • G.729a/b

SIP Server Support: Registrar Server (setting from web) • Outbound Proxy (setting from web)

IP Assignment: Static IP • DHCP • PPPoE

Security: HTTP 1.1 basic/digest • authentication for Web setup • MD5 for SIP authentication (RFC 2069/ RFC 2617)

QoS: ToS field • IEEE 802.1q VLAN • Tone • DTMF –(inband, out of band, SIP info) • 4 selectable ring tones • ring back tone (local & remote) • dial tone • busy tone

Dial Methods: Direct IP call without SIP registration • dial registered number via SIP server • dial URI from phone book/speed dial

Voice Quality: VAD (voice activity detection) • CNG (comfort noise generation) • AEC (acoustic echo cancellation) • G.168 • jitter buffer

Firmware Upgrade: TFTP • auto/manual provisioning system • NAT Traversal • UPnP • STUN • TCP/IP • IP/TCP/UDP/DHCP/RTP/RTCP • ICMP/HTTP/SNTP/TFTP/DNS

Configuration: Key & LCD configuration • web browser configuration • auto/manual provisioning system

Compliance: FCC Part 15 Class B • CE Class B • VCCI Class B • EN60950

I'm Brian, one of Patton's Product Validation Engineers who makes sure your VoIP product works reliably. To buy one of these state-of-the-art devices, call ***+1 301.975.1000*** *or send e-mail to* ***sales@patton.com***.

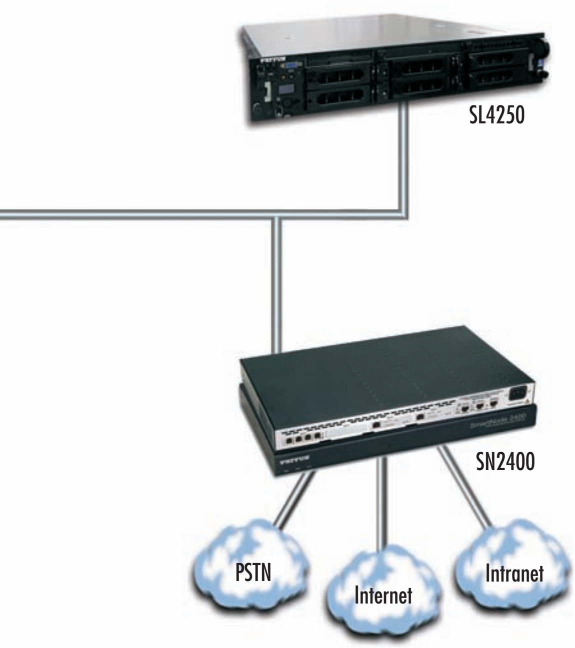
IP-PBX

SmartLink™ 4250 IP PBX

The Patton Smartlink 4250 IP PBX provides complete PBX services to VoIP equipment within centralized or decentralized enterprises.

The Patton IP PBX, the SmartNode family of analog and PSTN trunk gateways, and VoIP SIP phones are the elements of an enterprise PBX solution that can be purchased as complete turnkey system or as individual elements.

The IP PBX package is designed to serve as a replacement or augmentation to traditional telephony PBX systems while providing access to enhanced telephony features that are only available through VoIP. Single IP PBX systems are targeted for use in the private enterprise networks of up to 150 – 200 employees. Multiple systems can be implemented in larger organizations to provide seamless calling between systems or sites.



Web based administrative and user interfaces allow for quick and easy adjustment of calling preferences.

Enterprises operating in multiple locations can leverage their existing data intranet /VPN investment to extend the headquarters IP PBX all locations and telecommuters. Every employee can be reach on a simple 3 digit extension and all the features of the IP PBX are available to all employees. Intra-enterprise long-distance phones bills drop to zero while increasing organizational productivity. Additional cost savings can be achieved through downsizing of the local PSTN connection(s) and/or leveraging the local connections to implementing least cost telephony routing.

QoS capabilities on all elements of the Patton IP voice solution ensures that voice traffic can be given a higher priority than data within your network

The IP PBX includes a robust Voice mail system that allows users to retrieve their voice mails from any phone or interface with a browser. User preferences on the web interface allow each user to set voice message delivery options including delivery of voicemails to email.

The Smartlink 4250 is based on the VoIP industry standard SIP protocol (IETF RFC 3261) and provides compete SIP registration, subscription and notification services to any SIP capable end point. APIs in the software allow for easy integration of other 3rd party SIP based applications.

When partnered with a Patton Smartnode product, the Patton 4250 IP PBX can provide VoIP services to analog phones, faxes and legacy PBX systems that are not prepared to support newer VoIP technologies. Patton's ability to link voice technologies of the past with today's VoIP technologies allows your enterprise to cost effectively move forward.

FEATURES & BENEFITS

- ✔ Full Enterprise PBX, Call Control and Routing functions
- ✔ Uniquely leverages the unique functionality of Patton's Routers and Gateways & SIP Phones
- ✔ Standards-based implementation of the IETF Session Initiation Protocol (SIP) standard for end-to-end signaling.
- ✔ Simple to install with a browser-based configuration and management
- ✔ Easy to use with browser-based user interfaces
- ✔ Unprecedented customizable PBX system and unified messaging features
- ✔ Email notification and delivery of voice mail
- ✔ Scalable up to hundreds users—from 50 to 1000s
- ✔ Adaptable to a variety of telephony environments
- ✔ Modular architecture allows distributed network integration for multi-site, multi-station and multi-location support
- ✔ High reliability, backup and load balancing
- ✔ Highly manageable, productive, and secure
- ✔ Fully supported by Patton

I'm Dave, Patton's Manager of US Technical Support. If you have technical questions or comments, please call ***+1 301.975.1007****. You can also send e-mail to* ***support@patton.com****.*

VoIP Auto-Provisioning System (VAPS)

Provides centralized configuration and firmware control for large implementations of Patton VoIP Products

Service provider or enterprises with large deployments of Patton VoIP equipment require a way to centrally manage and control deployment of end user configurations and firmware. Configuration of individual devices via web interfaces is not a practical option for customers with a large installed base of end users. Patton's VAPS is a centralized provisioning and configuration tool that enables large scale roll-out, configuration, and firmware deployment of the Patton VoIP products at a device, group or network wide level.

Using web configuration screens similar to the web interface presented by the actual device, properly authenticated network operations personnel can review and adjust device configuration parameters. Changes that have been set to the VAPS database will be deployed to the selected devices the next time the device reloads or checks-in with VAPS server. Patton Smartnode and SmartLink products will check in with the VAPS server on a configuration defined interval.

The VAPS log tracks each VoIP device's check-in with the VAPS and what activities were performed. Network operators can quickly tell which devices received new configurations and new firmware.

To provide maximum connectivity to diverse end user environments, configuration and firmware is transmitted to and from VAPS using HTTP. Secure HTTPS is supported with the installation of an SSL public certificate on the VAPS server. Smartnode products offer the additional option of distributing configuration information using TFTP.

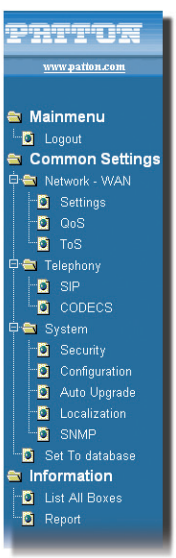
FEATURES & BENEFITS

- ✔ Centrally manages 100s–1,000s of Patton VoIP devices—Individual, group level or network-wide device configuration and firmware control
- ✔ Easy integration with existing Network Management Systems—Link to the VAPS database using OBDC
- ✔ Open standard HTTP or HTTPs-based data transfer—Provides maximum connectivity to end user devices located behind firewalls
- ✔ Eliminates the need manage to each device individually—device check-in deploys new firmware and software groups or all devices managed by VAPS
- ✔ Full integration with SmartLink and SmartNode products; Track device check-ins and upgrades—VAPS system log tracks firmware, configuration and check-in activity by device
- ✔ **VAPS Requirements**: Windows XP Server (XP Pro can be substituted for XP Server if fewer than 8 sessions will be served) with MS-SQL or Linux with MYSQL DB

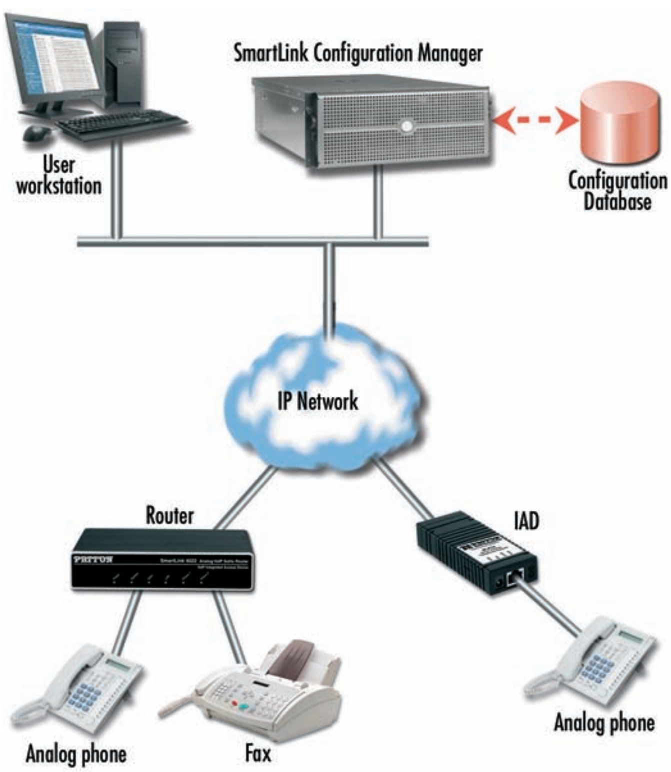
ORDERING INFORMATION

Call for a custom quote for your deployment.

VAPS device common menu

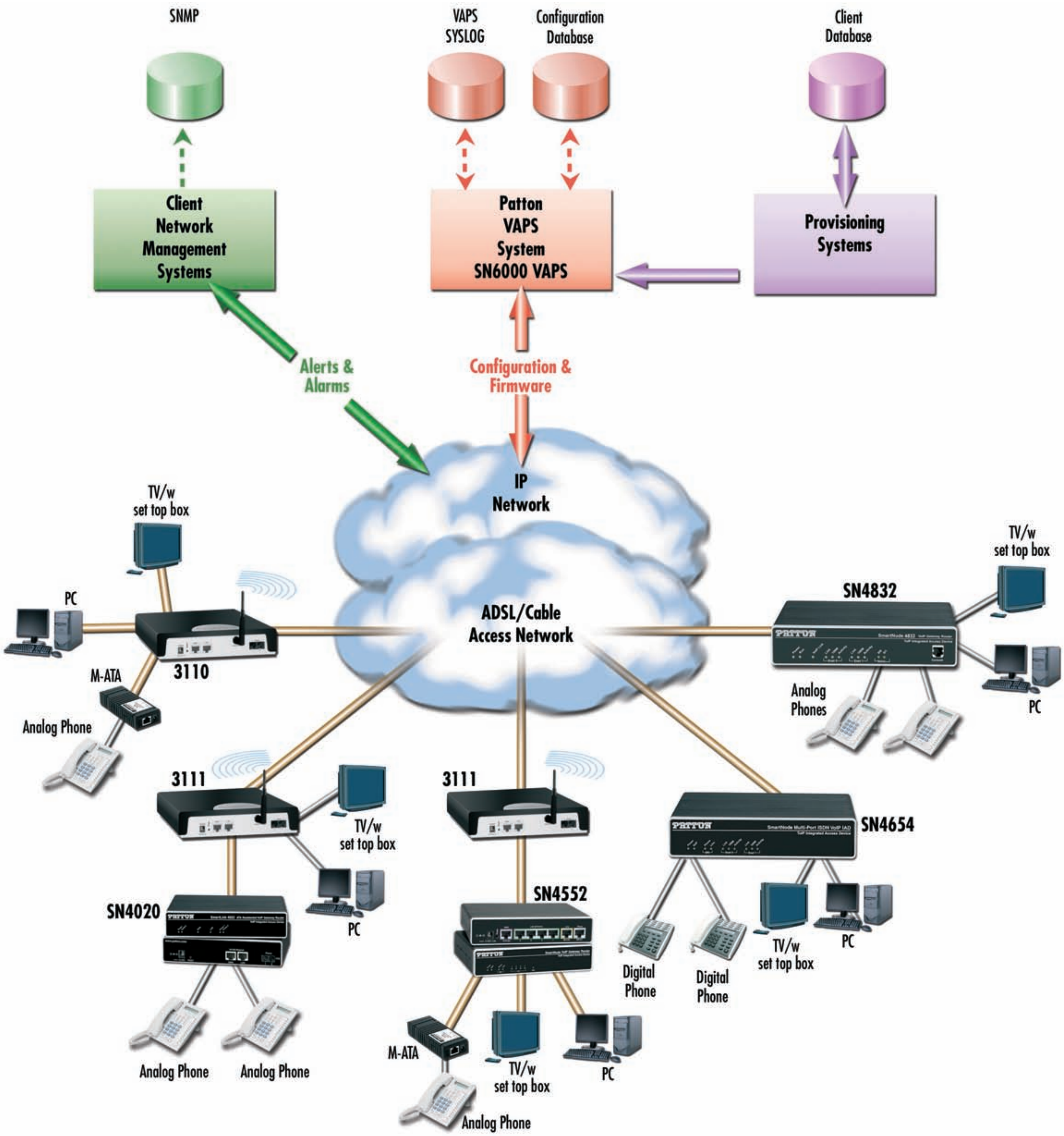


VAPS individual device configuration menu



- Manages SmartLink and SmartNode configurations with a common look-and-feel interface.
- Provides a mechanism through either push or pull to automatically update a single device, groups of devices or to make a change across the entire installed base of end user devices.
- Implements VAPS application access security with logging of configuration and access related activities to a syslog.
- Allows application of a group or system wide default configuration to a device or group of devices
- Implements secure transmission of the configuration to and from managed devices.

VAPS system diagram



VoIP Bundles

The Patton IP PBX, the SmartNode family of analog and PSTN trunk gateways and VoIP SIP phones are the elements of an enterprise PBX solution that can be purchased as complete turnkey system or as individual elements.

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Web based administrative and user interfaces allow for quick and easy adjustment of calling preferences.


SmartLink 4050/20


SmartNode Gateway


SIPxNano


SmartLink 4250

EXAMPLE PACKAGE

GS-40-1003 package includes:

Full Function IP PBX for 75 extensions
- Voice Mail
- Auto Attendant
- Voice to email integration
- Hunt Groups

PSTN Gateway
- Up to 24 concurrent PSTN calls
- Analog phone and FAX support

75 VoIP Phones
- 2 line speaker phones
- LCD display
- Local phone book
- Voicemail MWI
- Fax Support

Picking the right PBX package for your office is as easy as 1, 2, 3!

1. Pick your IP PBX		2. Pick your Gateway			3. Pick your Handsets and Complete Package			
Up to # of extensions	Then select your IP-PBX	# of Analog Phones or Faxes	# of PSTN Peak Calls	Then select your Gateway(s)	VoIP Phones	PBX, Gateway and VoIP phone package	Keep your existing Analog phones	PBX, Gateway and Analog phone package
15	SIPxNano-15	≤4	4	SN4528/4JS4JO	(15) SL4050/2	GS-40-1001	SN4912	GS-40-1011
30	SIPxNano-30	≤4	4	SN4528/4JS4JO	(30) SL4050/2	GS-40-1002	SN4932	GS-40-1012
75	SL4250-75	≤4	24	SN2400/1T1 w/(1) 4 FXS	(75) SL4050/2	GS-40-1003	(3) SN4924	GS-40-1013
			30	SN2400/1E1 w/(1) 4 FXS	(75) SL4050/2	GS-40-1004	(3) SN4924	GS-40-1014
125	SL4250-125	8	24	SN2400/1T1 w/(2) 4 FXS	(120) SL4050/2	GS-40-1005	(3) SN4932	GS-40-1015
			30	SN2400/1E1 w/(2) 4 FXS	(125) SL4050/2	GS-40-1006	(3) SN4932	GS-40-1016
250	SL4250-250	8	48	SN2400/2E1 w/(2) 4 FXS	(250) SL4050/2	GS-40-1007	(6) SN4932	GS-40-1017
			60	SN2400/2E1 w/(2) 4 FXS	(250) SL4050/2	GS-40-1008	(6) SN4932	GS-40-1018

Telecommuter/Remote office VoIP equipment selector

You can quickly extend all the capabilities of your office IP-PBX to your tele-workers and remote offices that have a dedicated Internet connection of at least 256 kbps.

	+ Local LAN		+ PSTN		
Number of Phones or Faxes	1	2	1	2	4
Then Pick →	M-ATA or M-AFA	SL4022	SN4522	SN4522	SN4524

Application diagram

ORDERING INFORMATION

GS-40-1001SipX: 15 User Package with 4xFXS 4xFXO Gateway; 15x 2-line SIP phones

GS-40-1011SipX: 15 User Package with 4xFXS 4xFXO Gateway; 12x FXS phone ports

GS-40-1002SipX: 30 User Package with 4xFXS 4xFXO Gateway; 30x 2-line SIP phones

GS-40-1012SipX: 30 User Package with 4xFXS 4xFXO Gateway; 32x 2-line SIP phones

GS-40-1003SL4250: 75 User Package with 1xT1 and 4xFXS Gateway; 75x 2-line SIP phones

GS-40-1013SL4250: 75 User Package with 1xT1 and 4xFXS Gateway; 72x FXS phone ports

GS-40-1004SL4250: 75 User Package with 1xE1 and 4xFXS Gateway; 75x 2-line SIP phones

GS-40-1014SL4250: 75 User Package with 1xE1 and 4xFXS Gateway; 72x FXS phone ports

GS-40-1005SL4250: 125 User Package with 1xT1 and 8xFXS Gateway; 120x 2-line SIP phones

GS-40-1015SL4250: 125 User Package with 1xT1 and 8xFXS Gateway; 72x FXS ports

GS-40-1006SL4250: 125 User Package with 1xE1 and 8xFXS Gateway; 120x 2-line SIP phones

GS-40-1016SL4250: 125 User Package with 1xE1 and 8xFXS Gateway; 72x FXS ports

GS-40-1007SL4250: 250 User Package with 1xT1 and 8xFXS Gateway; 250x 2-line SIP phones

GS-40-1017SL4250: 250 User Package with 1xT1 and 8xFXS Gateway; 144x FXS ports

GS-40-1008SL4250: 250 User Package with 1xE1 and 8xFXS Gateway; 250x 2-line SIP phones

GS-40-1018SL4250: 250 User Package with 1xE1 and 8xFXS Gateway; 144x FXS ports